sounds. Other metrics such as signal spectrum, duration, and stationarity may be considered in determining whether the ambient sound is analyzed for warning sounds. Notably, other metrics (e.g., context aware) may also be used to determine when the ambient sound should be processed for warning sound detection.

[0070] If at least one property (e.g., power, spectral shape, duration, etc.) of the ambient sound exceeds a threshold (or adaptive threshold), the processor 114 at step 530 can proceed to generate a sound signature for the ambient sound. In one embodiment the sound signature is a feature vector which can include statistical parameters or salient features of the ambient sound.

[0071] An ambient sound with a warning sound (e.g. "bell", "siren"), such as shown in step 532, is generally expected to exhibit features similar to sound signatures for similar warning sounds (e.g. "bell", "siren") stored in the user defined database 504 or the on-line database 506. The processor 114 can also identify a direction and speed of the sound source if it is moving, for example, by evaluating Doppler shift as shown in step 534 and 536. The processor 114, by way of beam-forming among multiple ASMs 201 may also estimate a direction of a sound source generating the warning sound.

[0072] The speed and bearing of the sound source can also be estimated using pitch analysis to detect changes predicted by Doppler effect, or alternatively by an analysis in changes in relative phase and magnitude between the two ASM signals. The processor 114, by way of a sound recognition engine, may detect general warning signals such as car horns or emergency sirens (and other signals referenced by ISO 7731) using spectral and temporal analysis.

[0073] The processor 114 can also analyze the ambient sound to determine if a verbal warning (e.g. "help", "police", "excuse me") is present. As shown in step 540, the sound signature of the ambient sound can be analyzed for speech content. For example, the sound signature may be analyzed for voice information, such as vocal cord pitch periodicities, time-varying voice formant envelopes, or other articulation parameter attributes. Upon detecting the presence of voice in the ambient sound, the processor 114 can perform key word detection (e.g. "help") in the spoken content as shown in step 542. Speech recognition models as well as language models may be used to identify key words in the spoken content. As previously noted, the user 124 may say or enter in one or more warning sounds that may be mapped to associated learning models for sound signature detection.

[0074] As shown in step 552, the user 124 may also provide user input to direct operation of the processor 114, for example, to select an operational mode as shown in 550. As one example, the operational mode can enable, disable or adjust monitoring of warning sounds. For example, in a listening mode, the processor 114 may mix audio

content with ambient sound while monitoring for warning sounds. In a quiet mode, the processor 114 may suppress or attempt to actively cancel all noises except detected warning sounds.

[0075] The user input may be in the form of a physical interaction (e.g., button press) or a vocalization (e.g., spoken command). The operating mode can also be controlled by a prioritizing module as shown in step 554. The prioritizing module may prioritize warning sounds based on severity and context. For example, if the user 124 is in a phone call, and a warning sound is detected, the processor 114 may audibly inform the user 124 of the warning and/or present a text message of the warning sound. If the user 124 is listening to music or a voice communication, and a warning sound is detected, the processor 114 may automatically shut off the music or voice audio and alert the user. The user 124, by way of user interface 106 (FIG. 1) or an administrator, may rank warning sounds and instruct the processor 114 how to respond to warnings in various contexts.

[0076] FIG. 6 is a flowchart of a method 600 for managing audio delivery based on detected sound signatures in accordance with an exemplary embodiment. The method 600 can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method 600, reference will be made to components of FIG 2 , although it is understood that the method 600 can be implemented in any other manner using other suitable components.

[0077] As noted previously, the audio interface 212 can supply audio content (e.g., music, cell phone, voice mail, etc.) to the processor 114. In such regard, the user 124 may listen to music, talk on the phone, receive voice mail, or perform other audio related tasks while the processor 114 additionally monitors warning sounds in the environment. During normal use, when a warning sound is not present, the processor 114 may operate normally to recreate the sound experience requested by the user 124. If however the processor 114 detects a warning sound, the processor 114 may manage audio content delivery to notify the user 124 of the warning sound. Managing audio content delivery can include adjusting or overriding other current audio settings.

[0078] By way of example, as shown in step 602, the audio interface 212 receives audio content from a media player, such as a portable music player, or cell phone. The audio content can be delivered to the user's vehicle cabin by way of the VCRs 112, 120 as shown in step 604.

[0079] At step 606, the processor 114 monitors ambient sound in the environment captured at the ASM 201. Ambient sound may be sampled at sufficiently data rates (e.g. 8 kHz, 16 kHz, and 32 KHz) to allow for feature extraction of sound signatures. Moreover, the processor 114 may adjust the sampling rate based on the information

content of the ambient signal. For example, upon the ambient sound exceeding a first threshold, the sampling rate may be set to a first rate (e.g., 4 KHz). As the ambient sound increases in volume, or as prominent features are identified, the sampling rate may be increased to a second rate (e.g., 8 KHz) to increase signal resolution. Although, the higher sampling rate may improve resolution of features, the lower sampling rate may preserve use of computational resources for minimally sufficient feature resolution (e.g., power supply 210, processor 114).

[0080] If at step 608, a sound signature is detected, the processor 114 may determine a priority of the detected sound signature (at step 610). The priority establishes how the processor 114 manages audio content. Notably, warning sounds for various environmental conditions and user experiences can be learned. Accordingly, the user 124 or an administrator, can establish priorities for warning sounds. Moreover, these priorities may be based on environmental context. For example, if a user 124 is in a warehouse where loading vehicles emit a beeping sound, sound signatures for such vehicles can be given the highest priority. A user 124 may also prioritize learned warning sounds, for example, via a user interface on a paired device (e.g., cell phone), or via speech recognition (e.g., "prioritize – 'ambulance' – high").

[0081] Upon detecting a warning sound and identifying a priority, the processor 114, at step 612, selectively manages at least a portion of the audio content based on the priority. For example, if the user 124 is listening to music during the time a warning sound is detected, the processor 114 may decrease the music volume to present an audible notification. This may represent one indication that the processor 114 has detected a warning sound.

[0082] At step 614, the processor 114 may further present an audible notification to the user 124. For example, upon detecting a "horn" sound, a speech-to-text message can be presented to the user 124 to audibly inform them that a horn sound has been detected (e.g., "horn detected"). Information related to the warning sound (e.g., direction, speed, priority, etc.) may also be presented with the audible notification.

[0083] In a further arrangement, the processor 114 may send a message to a device operated by the user 124 to visually display the notification, as shown in step 616. For example, if the user has disengaged audible notification, the processor 114 may transmit a text message to a paired device (e.g., a cell phone) containing the audible warning (or to indicator 116 configured as a visual indicator). Moreover, the processor 114 may beacon out an audible alarm to other devices within a vicinity, for example via Wi-Fi (e.g., IEEE 802.16x). Other devices in a proximity of the user 124 may sign up to receive audible alarms from the processor 114. In such regard, the processor 114 can

beacon a warning notification to other devices in the area to share warning information with other people.

[0084] FIG. 7 is a flowchart of a method 700 further describing sound signature detection in accordance with an exemplary embodiment. The method 700 can be practiced with more or fewer than the number of steps shown and is not limited to the order shown. The method 700 can begin in a state in which the processor 114 is actively monitoring warning sounds in the environment.

[0085] At step 711, ambient sound captured from the ASM 201 may be buffered into short term memory as frames. As an example, the ambient sound may be sampled at 8 KHz with 10-20 ms frame sizes (80 to 160 samples). The frame size may also vary depending on the energy level of the ambient sound. For example, the processor 114 upon detecting low level sounds (e.g., about 70-74 dB SPL) may use a frame size of about 30 ms, and update the frame size to about 10 ms as the power level increases (e.g., greater than about 86 dB SPL). The processor 114 may also increase the sampling rate in accordance with the power level and/or a duration of the ambient sound. (A longer frame size with lower sampling may compromise resolution for computational resources.) The data buffer is desirably of sufficient length to hold a history of frames (e.g., about 10-15 frames) for short-term historical analysis.

[0086] At step 712, the processor 114 may perform feature extraction on the frame as the ambient sound is buffered into the data buffer. As one example, feature extraction may include performing a filter-bank analysis and summing frequencies in auditory bandwidths. Features may also include Fast Fourier Transform (FFT) coefficients, Discrete Cosine Transform (DCT) coefficients, cepstral coefficients, partial autocorrelation (PARCOR) coefficients, wavelet coefficients, statistical values (e.g., energy, mean, skew, variance), parametric features, or any other suitable data compression feature set.

[0087] Additionally, dynamic features, such as derivatives of any order, may be added to the static feature set. As one example, mel-frequency-cepstral analysis may be performed on the frame to generate between about 10-16 mel-frequency-cepstral coefficients. The small number of coefficients represent features that may be compactly stored to memory for that particular frame. Such front end feature extraction techniques may reduce the amount of data used to represent the data frame.

[0088] At step 713, the features may be incorporated as a sound signature and compared to learned models, for example, those retrieved from the warning sounds database 718 (e.g., user defined database 504 or the on-line database 506 of FIG. 5). A sound signature may be defined as a sound in the user's ambient environment which has significant perceptual saliency. As an example, a sound signature may correspond to an

alarm, an ambulance, a siren, a horn, a police car, a bus, a bell, a gunshot, a window breaking, or any other warning sound, including voice. The sound signature may include features characteristic to the sound. As an example, the sound signature may be classified by statistical features of the sound (e.g., envelope, harmonics, spectral peaks, modulation, etc.).

[0089] Notably, each learned model used to identify a sound signature has a set of features specific to a warning sound. For example, a feature vector of a learned model for an "alarm" is sufficiently different from a feature vector of a learned model for a "bell sound". Moreover, the learned model may describe interconnectivity (e.g., state transitions, emission probabilities, initial probabilities, synaptic connections, hidden layers) among the feature vectors (e.g. frames). For example, the features of a "bell" sound may change in a specific manner compared to the features of an "alarm" sound. The learned model may be a statistical model such as a Gaussian mixture model (GMM), a Hidden Markov Model (HMM), a Bayes Classifier, or a Neural Network (NN) that requires training.

[0090] In the foregoing, a Gaussian Mixture Model (GMM) is presented, although it should be noted that any of the above models may be used for sound signature detection. In this case, each warning sound may have an associated GMM used for detecting the warning sound. As an example, the warning sound for an "alarm" may have its own GMM, and a warning sound for a "bell" may have its own GMM. Separate GMMs may also be used as a basis for the absence of the sounds ("anti-models"), such as "not alarm" or "not bell." Each GMM may provide a model for the distribution of the feature statistics for each warning sound in a multi-dimensional space.

[0091] Upon presentation of a new feature vector, the likelihood of the presence of each warning sound may then be calculated. In order to detect a warning sound, each warning sound's GMM may be evaluated relative to its anti-model, and a score related to the likelihood of that warning sound may be computed. A threshold may be applied directly to this score to decide whether the warning sound is present or absent. Similarly, the sequence of scores may be relayed to yet another module which uses a more complex rule to decide presence or absence. Examples of such rules include linear smoothing or median filtering.

[0092] As previously noted, a HMM model or NN model with their associated connection logic may be used in place of each GMM for each learning model. For example, each warning sound in the database 718 may have a corresponding HMM. A sound signature for a warning sound captured at the ASM 201 in ambient sound may be processed through a lattice network (e.g. Viterbi network) for comparison to each HMM, to determine which HMM, if any, corresponds to the warning sound. Alternatively, in a

trained NN, the sound signature may be input to the NN, where the output states of the NN correspond to warning sound indices. The NN may include various topologies such as a Feed-Forward, Radial Basis Function, Hopfield, Time-Delay Recurrent, or other optimized topologies for real-time sound signature detection.

[0093] At step 714, a distortion metric is performed with each learned model to determine which learned models are closest to the captured feature vector (e.g., sound signature). The learned model with the smallest distortion (e.g., mathematical distance) is generally considered the correct match, or recognition result. It should also be noted that the distortion may be calculated as part of the model comparison in step 713. This is because the distortion metric may depend on the type of model used (e.g., HMM, NN, GMM, etc.) and in fact may be internal to the model (e.g. Viterbi decoding, back-propagation error update, etc.). The distortion module is merely presented in FIG. 7 as a separate component to suggest use with other types of pattern recognition methods or learning models.

[0094] Upon evaluating the feature vector (e.g. sound signature) against the candidate warning sound learned models, the ambient sound at step 715 may be classified as a warning sound. Each of the learned models may be associated with a score. For example, upon the presentation of a sound signature, each GMM may produce a score. The scores may be evaluated against a threshold, and the GMM with the highest score may be identified as the detected warning sound. For example, if the learned model for the "alarm" sound produces the highest score (e.g., smallest distortion result) compared to other learned models, the ambient sound may be classified as an "alarm" warning sound.

[0095] The classification step 715 also takes into account likelihoods (e.g. recognition probabilities). For example, as part of the step of comparing the sound signature of the unknown ambient sound against all the GMMs for the learned models, each GMM may produce a likelihood result, or output. As an example, these likelihood results may be evaluated against each other or in a logical context, to determine the GMM considered "most likely" to match the sound signature of the warning sound. The processor 114 may then select the GMM with the highest likelihood or score via soft decisions.

[0096] The processor 114 may continually monitor the environment for warning sounds, or monitor the environment on a scheduled basis. In one arrangement, the processor 114 may increase monitoring in the presence of high ambient noise possibly signifying environmental danger or activity. Upon classifying an ambient sound as a warning sound the processor 114, at step 716, the ASRS 100 may generate an alarm. As previously noted, the processor 114 may mix the warning sound with audio content, amplify the warning sound, reproduce the warning sound, and/or deliver an audible

message. As one example, spectral bands of the audio content that mask the warning sound may be suppressed to increase an audibility of the warning sound. This serves to notify the user 124 of a warning sounded detected in the environment, of which the user 124 may not be aware of depending on their environmental context.

[0097] As an example, the processor 114 may present an amplified audible notification to the user via the VCRs 112, 120. The audible notification may be a synthetic voice identifying the warning sound (e.g. "car alarm"), a location or direction of the sound source generating the warning sound (e.g. "to your left"), a duration of the warning sound (e.g., "3 minutes") from initial capture, and any other information (e.g., proximity, severity level, etc.) related to the warning sound. Moreover, the processor 114 may selectively mix the warning sound with the audio content based on a predetermined threshold level. For example, the user 124 may prioritize warning sound types for receiving various levels of notification, and/or identify the sound types as desirable of undesirable.

[0098] FIG. 8, presents a pictorial diagram 800 for mixing ambient sounds and warning sounds with audio content. In the illustration show, the processor 114 is directing music 136 to the vehicle cabin 126 via VCR 120 while simultaneously monitoring warning sounds in the environment. At time, T, the processor 114, upon detecting a warning sound (signature 135), can lower the music volume from the media player 150 (graph 141), and increase the volume of the ambient sound received at the ASM 201 (graph 142). Other mixing arrangements are herein contemplated.

[0099] In such regard, there is a smooth audio transition between the music 136 and the warning sound 135. Notably, the ramp up and down times can also be adjusted based on the priority of the warning sound. For example, in an extreme case, the processor 114 may immediately shut off the music, and present the audible warning. Other various implementations for mixing audio and managing audio content delivery are herein contemplated.

[00100] Moreover, the audio content may be managed with other media devices (e.g., a cell phone). For example, upon detecting a warning sound, the processor 114 may inform the user 124 and the called party of a warning sound. In such regard, the user 124 does not need to inform the called party since they also receive the notification, which may save them time to explain an emergency situation.

[00101] As one example, the processor 114 may spectrally enhance the audio content in view of the ambient sound. Moreover, a timbral balance of the audio content may be maintained by taking into account level dependent equal loudness curves and other psychoacoustic criteria (e.g., masking) associated with a personalized hearing level (PHL) 430. For example, auditory queues in a received audio content may be enhanced

based on the PHL 430 and a spectrum of the ambient sound captured at the ASM 201. Frequency peaks within the audio content may be elevated relative to ambient noise frequency levels and in accordance with the PHL 430 to permit sufficient audibility of the ambient sound. The PHL 430 reveals frequency dynamic ranges that may be used to limit the compression range of the peak elevation in view of the ambient noise spectrum.

[00102] In one arrangement, the processor 114 may compensate for a masking of the ambient sound by the audio content. Notably, the audio content, if sufficiently loud, may mask auditory queues in the ambient sound, which can: i) potentially cause hearing damage, and ii) prevent the user 124 from hearing warning sounds in the environment (e.g., an approaching ambulance, an alarm, etc.). Accordingly, the processor 114 may accentuate and attenuate frequencies of the audio content and ambient sound to permit maximal sound reproduction while simultaneously permitting audibility of ambient sounds.

[00103] In one exemplary embodiment, the processor 114 may narrow noise frequency bands within the ambient sound to permit sensitivity to audio content between the frequency bands. The processor 114 may also determine if the ambient sound contains salient information (e.g., warning sounds) that should be un-masked with respect to the audio content. If the ambient sound is not relevant, the processor 114 may mask the ambient sound (e.g., increase levels) with the audio content until warning sounds are detected.

[00104] FIG. 9 is a flowchart of a method 900 for updating the sound signature detection library dependent on the vehicle location. At step, 902, a current vehicle location may be acquired. The location of vehicle 102 may be determined with a number of methods, including: a Global Positioning System (GPS) for determining a GPS location and cell-phone signal cell codes for triangulating the location (for example, based on a signal strength of the received signal(s) from cell phone networks).

[00105] The acquired vehicle location may be used, at step 904, to determine which "sound library cell" the vehicle 102 is located in. The sound library cell may refer to a geographic region, such as a country, state, province or city. The sound library cell may contain the target sound signatures which the vehicle's sound recognition system detects (for example, for a particular sound signature of a particular fire engine siren, there may be associated GMM statistics, as described above). The cell library may be determined from the acquired location (for example, using a look-up table), or may be acquired directly from coded signals embedded in the received location signal (for example, a cell-phone signal or a radio signal).

[00106] If, at step 906, the vehicle library cell location is determined to have changed, then the current sound signature library of the vehicle is updated (at step 910). For

example, the current vehicle sound signature library may be updated from data provided by an on-line sound signature library 908, where the new sound signature data may be received wirelessly or from a storage device located on the vehicle 102.

[00107] Although the invention has been described in terms of automotive sound recognition systems and methods for enhancing situation awareness in a vehicle, it is contemplated that one or more steps and/or components may be implemented in software for use with microprocessors/general purpose computers (not shown). In this embodiment, one or more of the functions of the various components and/or steps described above may be implemented in software that controls a computer. The software may be embodied in non-transitory tangible computer readable media (such as, by way of non-limiting example, a magnetic disk, optical disk, flash memory, hard drive, etc.) for execution by the computer.

[00108] Although the invention is illustrated and described herein with reference to specific embodiments, the invention is not intended to be limited to the details shown. Rather, various modifications may be made in the details within the scope and range of equivalents of the claims and without departing from the invention.

What is Claimed:

1. A sound recognition system for a vehicle comprising:

at least one ambient sound microphone (ASM), disposed on the vehicle, configured to capture ambient sound external to the vehicle;

at least one vehicle cabin receiver (VCR) configured to deliver audio content to a vehicle cabin of the vehicle; and

a processor, coupled to the at least one ASM and the at least one VCR, the processor configured to detect at least one sound signature in the ambient sound and to adjust the audio content delivered to the vehicle cabin based on the detected at least one sound signature.

2. The system according to claim 1, wherein the sound signature includes at least one of a non-verbal warning sound or a verbal warning sound.

3. The system according to claim 2, wherein the non-verbal warning sound includes at least one of an alarm, a horn, a siren or a noise.

4. The system according to claim 2, wherein the verbal warning sound includes one or more spoken words associated with a verbal warning.

5. The system according to claim 1, wherein the processor is configured to selectively adjust a volume of the audio content delivered to the vehicle cabin when the at least one sound signature is detected.

6. The system according to claim 1, wherein the processor is configured to reproduce the ambient sound associated with the at least one sound signature within the vehicle cabin.

7. The system according to claim 1, wherein the processor is configured to selectively mix the audio content with the ambient sound when the at least one sound signature is detected.

8. The system according to claim 1, further including a memory configured to store a target sound captured by the at least one ASM for learning the corresponding sound signature.

9. The system according to claim 8, wherein the memory is configured to store the target sound responsive to an indication by a user of the system.

10. The system according to claim 8, wherein the memory is configured to store the target sound automatically by the system.

11. The system according to claim 1, further including an audio interface coupled to the processor to receive the audio content from at least one of a media player or a mobile phone.

12. The system according to claim 1, wherein the system is configured to transmit one or more of the ambient sound and the at least one sound signature to a remote location.

13. A method for increasing auditory situation awareness in a vehicle, the method comprising the steps of:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle;

monitoring the ambient sound for a target sound by detecting a sound signature corresponding to the target sound in the ambient sound; and

adjusting a delivery of audio content by at least one vehicle cabin receiver (VCR) to a vehicle cabin of the vehicle based on the target sound.

14. The method according to claim 13, wherein the adjusting of the delivery of the audio content includes mixing the target sound with the audio content for delivery to the vehicle cabin.

15. The method according to claim 14, wherein the target sound is mixed with the audio content in accordance with a priority of the target sound.

16. The method according to claim 13, wherein the adjusting of the delivery of the audio content includes at least one of:

passing the target sound to the at least one VCR,

amplifying the target sound for delivery to the vehicle cabin,

attenuating the target sound for delivery to the vehicle cabin,

generating an audible message based on the target sound for delivery to the vehicle cabin, or

replacing the target sound with a predetermined sound corresponding to the target sound for delivery to the vehicle cabin.

17. The method according to claim 13, the method further including:

detecting at least one of a direction of a sound source or a speed of the sound source generating the target sound from the sound signature; and

indicating the at least one of the direction or the speed of the sound source in the vehicle cabin.

18. The method according to claim 13, the method further including transmitting a warning notification to other devices.

19. The method of claim 13, wherein the target sound includes at least one of an alarm, a horn, a siren, a spoken utterance or a noise.

20. The method according to claim 13, wherein the detecting of the sound signature includes detecting a spoken utterance in the ambient sound associated with a verbal warning, the method further including:

indicating the verbal warning in the vehicle cabin.

21. The method according to claim 13, the method further including:

acquiring a current location of the vehicle;

associating the current location with a sound signature; and

updating a sound signature library containing a plurality of predetermined target sounds with the sound signature associated with the current location.

22. A method for sound signature detection for a vehicle, the method comprising:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle; and

receiving a directive to learn a sound signature within the ambient sound,

wherein a voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.

23. The method according to claim 22, further including saving the sound signature at least one of locally on the vehicle or remotely to a server.

24. The method according to claim 22, further including adapting a previously learned warning sound model using the sound signature within the ambient sound.

25. A method for personalized listening in a vehicle, the method comprising:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle;

detecting a sound signature within the ambient sound that is associated with a warning sound; and

mixing the warning sound with audio content delivered to the vehicle cabin via at least one vehicle cabin receiver (VCR) in accordance with a priority of the warning sound and a personalized hearing level (PHL).

26. The method according to claim 25, wherein the detecting of the sound signature includes:

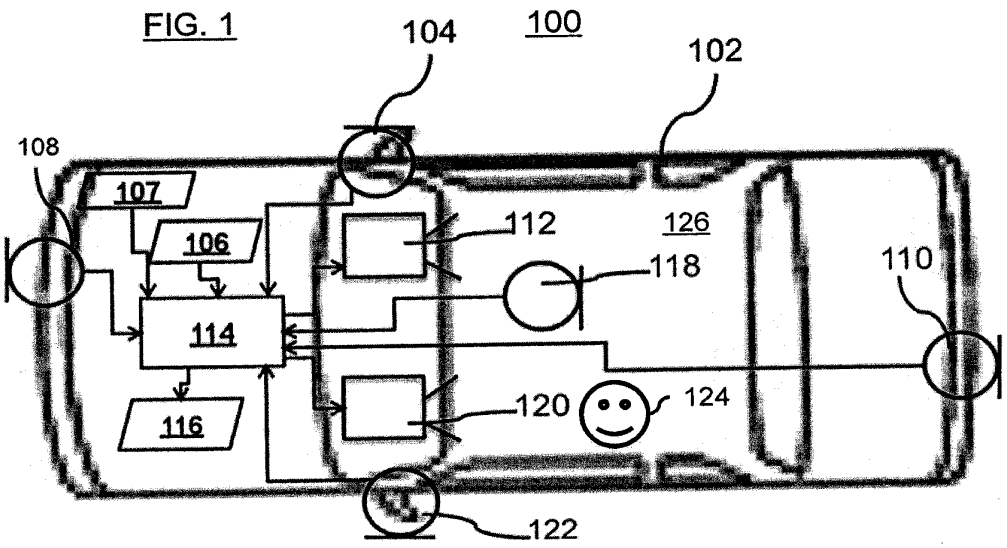
retrieving learned models from a database;

comparing the sound signature to the learned models; and

identifying the warning sound from the learned models responsive to the comparison.

27. The method according to claim 25, further including enhancing auditory queues in the warning sound relative to the audio content based on a spectrum of the ambient sound captured at the at least one ASM.

FIG. 1


104
100
102
108
107
106
112
126
118
110
114
116
120
124
122

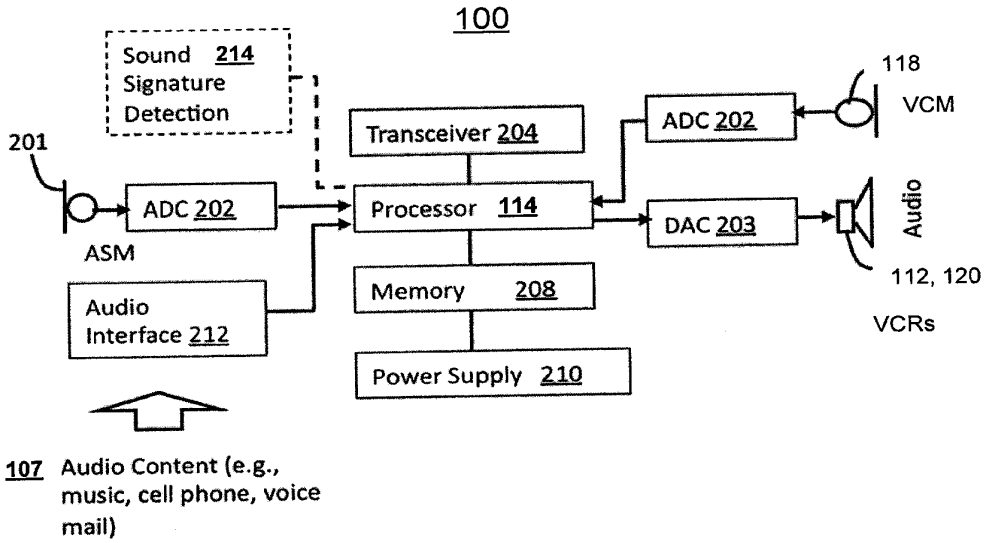

100
Sound 214
Signature
Detection
Transceiver 204
ADC 202
118
VCM
201
ADC 202
Processor 114
DAC 203
Audio
ASM
112, 120
Memory 208
Audio
Interface 212
VCRs
Power Supply 210
107 Audio Content (e.g., music, cell phone, voice mail)

FIG. 2

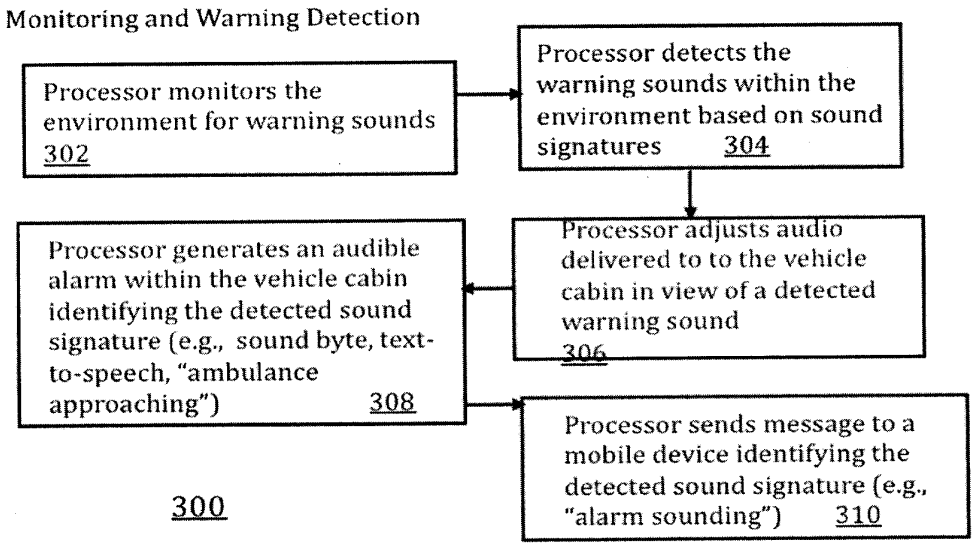

Monitoring and Warning Detection
Processor monitors the environment for warning sounds 302
Processor detects the warning sounds within the environment based on sound signatures 304
Processor adjusts audio delivered to to the vehicle cabin in view of a detected warning sound 306
Processor generates an audible alarm within the vehicle cabin identifying the detected sound signature (e.g., sound byte, text-to-speech, "ambulance approaching") 308
Processor sends message to a mobile device identifying the detected sound signature (e.g., "alarm sounding") 310
300

FIG. 3

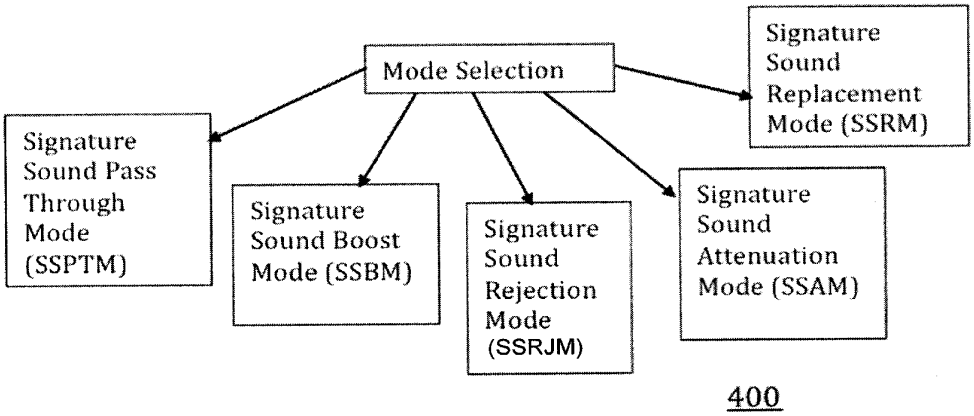

Mode Selection
Signature Sound Pass Through Mode (SSPTM)
Signature Sound Boost Mode (SSBM)
Signature Sound Rejection Mode (SSRJM)
Signature Sound Attenuation Mode (SSAM)
Signature Sound Replacement Mode (SSRM)
400

FIG. 4

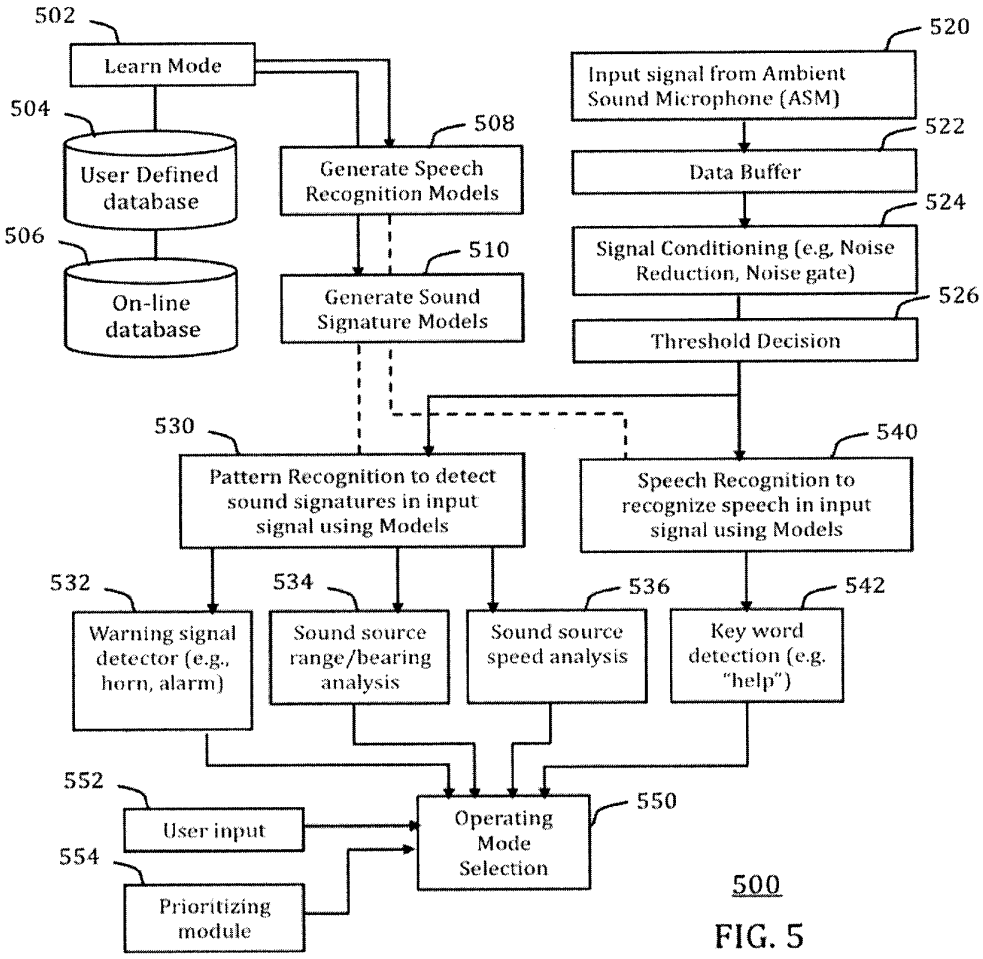

502
Learn Mode
504
User Defined database
506
On-line database
508
Generate Speech Recognition Models
510
Generate Sound Signature Models
520
Input signal from Ambient Sound Microphone (ASM)
522
Data Buffer
524
Signal Conditioning (e.g. Noise Reduction, Noise gate)
526
Threshold Decision
530
Pattern Recognition to detect sound signatures in input signal using Models
540
Speech Recognition to recognize speech in input signal using Models
532
Warning signal detector (e.g., horn, alarm)
534
Sound source range/bearing analysis
536
Sound source speed analysis
542
Key word detection (e.g. "help")
552
User input
554
Prioritizing module
550
Operating Mode Selection
500

FIG. 5

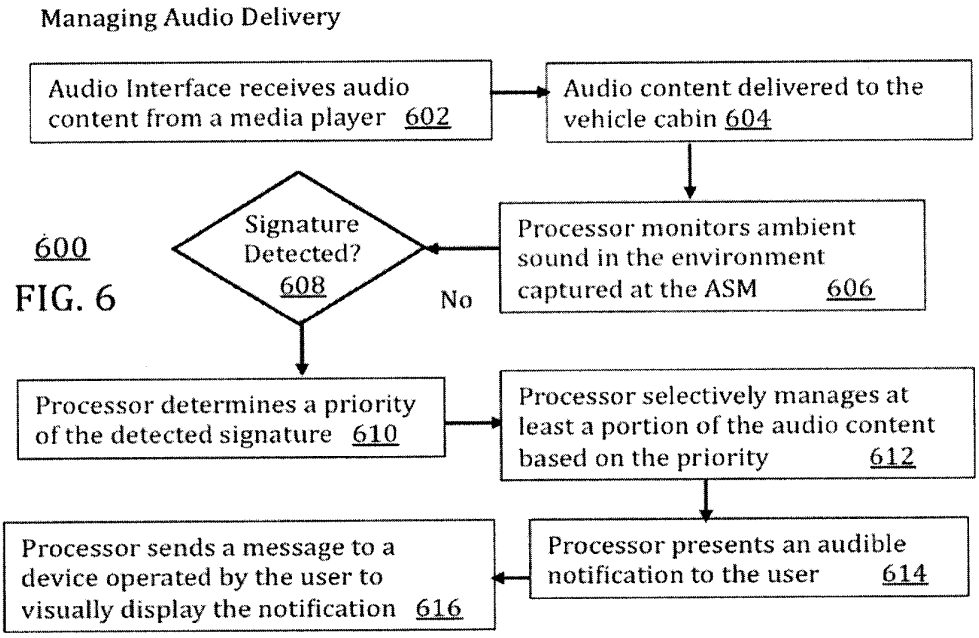

Managing Audio Delivery
Audio Interface receives audio content from a media player 602
Audio content delivered to the vehicle cabin 604
Processor monitors ambient sound in the environment captured at the ASM 606
Signature Detected? 608
No
Processor determines a priority of the detected signature 610
Processor selectively manages at least a portion of the audio content based on the priority 612
Processor presents an audible notification to the user 614
Processor sends a message to a device operated by the user to visually display the notification 616
600

FIG. 6

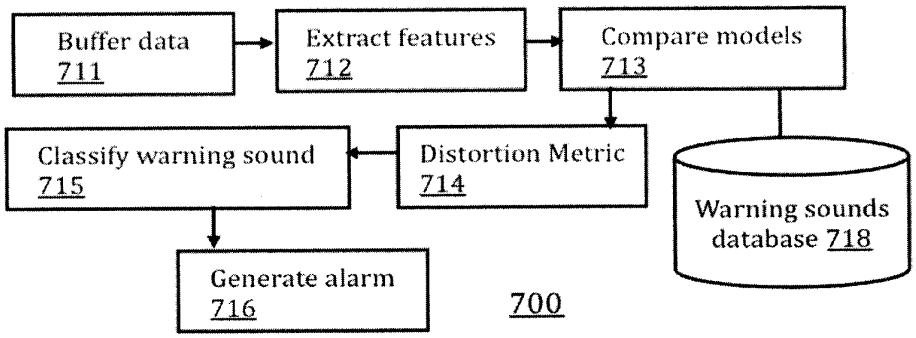
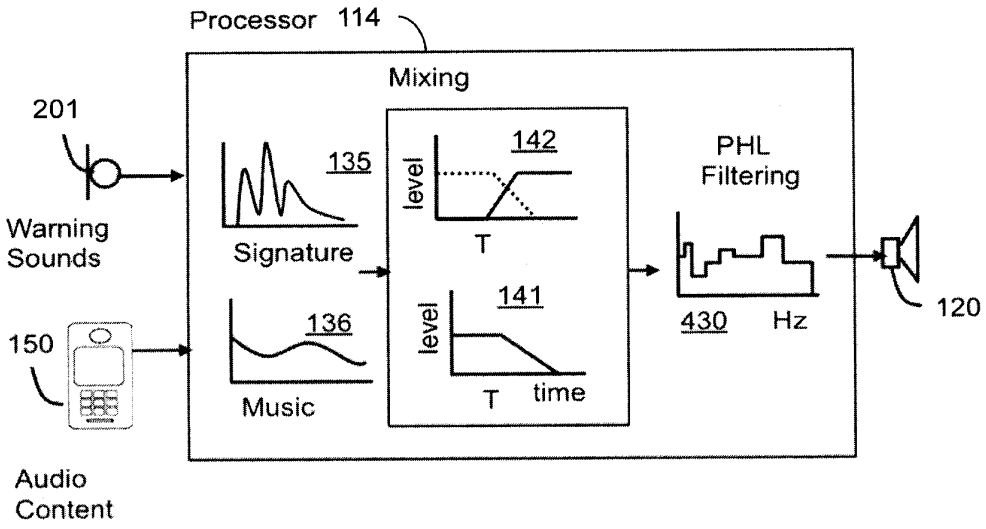

Buffer data 711
Extract features 712
Compare models 713
Classify warning sound 715
Distortion Metric 714
Warning sounds database 718
Generate alarm 716
700

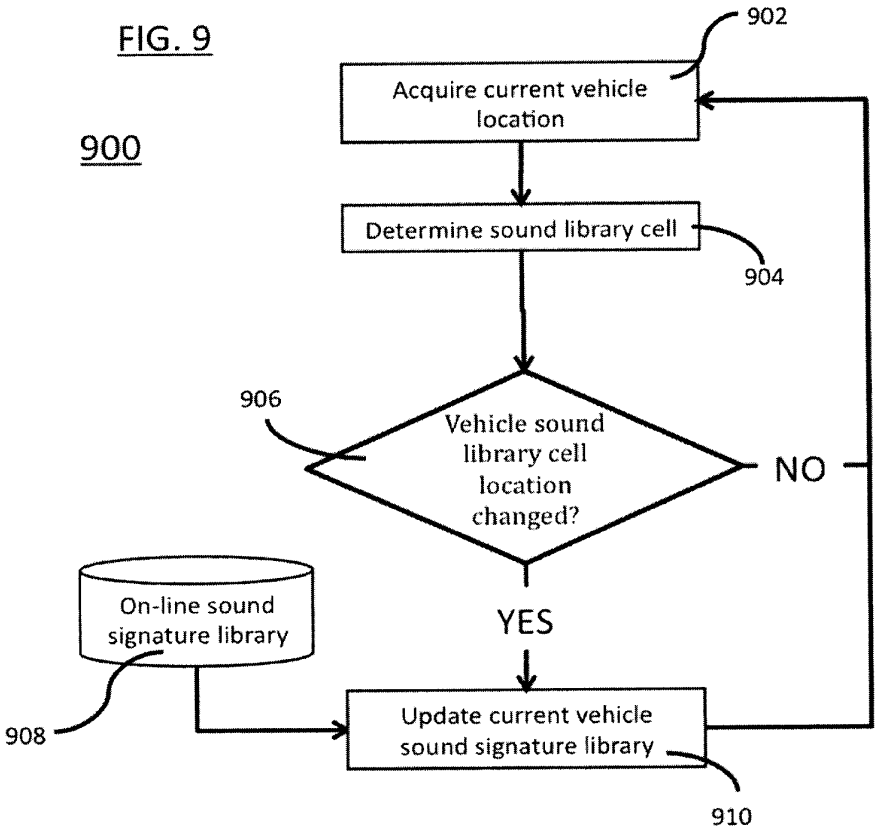
FIG. 7


Processor 114
Mixing
201
Warning Sounds
135
Signature
142
level
T
PHL Filtering
150
136
Music
141
level
T
time
430 Hz
120
Audio Content

800

FIG. 8

FIG. 9


900
902
Acquire current vehicle location
Determine sound library cell
904
906
Vehicle sound library cell location changed?
NO
YES
On-line sound signature library
908
Update current vehicle sound signature library
910

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US 12/21077

A. CLASSIFICATION OF SUBJECT MATTER
IPC(8) - H04B 1/00 (2012.01)
USPC - 381/86
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
USPC 381/86

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
USPC 381/86, 87, 91, 94.1; 180/167

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
PubWEST (PGPB, USPT, EPAB, JPAB); Google.com; Google Scholar.
Search terms: vehicle, automobile, external microphone, ambient, exterior, noise, sound, warning, alarm, horn, siren, voice, volume, music, sound signature, verbal, spoken.

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 2007/0127734 A1 (Brulle-Drews) 07 June 2007 (07.06.2007) Entire document, especially Title, para. [0006], [0037], Fig. 4, 5.	1-27
Y	US 2008/0240458 A1 (Goldstein et al.) 2 October 2008 (02.10.2008) Entire document, especially para. [0002], [0009], [0011], [0026], [0028], [0036], [0040], [0045]-[004], [0052], [0059], [0061], [0065], Fig. 2, 8.	1-27

☐ Further documents are listed in the continuation of Box C. ☐

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Date of the actual completion of the international search	Date of mailing of the international search report
19 March 2012 (19.03.2012)	26 MAR 2012
Name and mailing address of the ISA/US Mail Stop PCT, Attn: ISA/US, Commissioner for Patents P.O. Box 1450, Alexandria, Virginia 22313-1450 Facsimile No. 571-273-3201	Authorized officer: Lee W. Young PCT Helpdesk: 571-272-4300 PCT OSP: 571-272-7774

Electronic Acknowledgement Receipt

EFS ID:	46099327
Application Number:	17736180
International Application Number:	
Confirmation Number:	5894
Title of Invention:	PERSONALIZED SOUND MANAGEMENT AND METHOD
First Named Inventor/Applicant Name:	Steven Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-17203731
Receipt Date:	01-JUL-2022
Filing Date:	04-MAY-2022
Time Stamp:	11:04:03
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Transmittal Letter	IDS-transmittal-ltr.pdf	121529 c86acced709a845a9757f123989fe26f68651664	no	1

Warnings:

Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARDWARE-IDS-UPDATED.pdf	1036546 c78092031caf71a3d2a3f792ab620d58d6f3c609	no	11
Warnings:					
Information:					
3	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS-UPDATED.pdf	1036977 abfb638fdd26c6c65ceb022f8df5daf8bc8892b5	no	12
Warnings:					
Information:					
4	Information Disclosure Statement (IDS) Form (SB08)	MASTER-FOREIGN-NONPATENT-IDS.pdf	1035430 f201fab961b8821129c3c19f5a7491a91bac9aa8	no	5
Warnings:					
Information:					
5	Information Disclosure Statement (IDS) Form (SB08)	MASTER-IPR-IDS.pdf	1034856 a4ccccdf0dd46e883fd8c4caa3967b0845059bbe2	no	5
Warnings:					
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7	Foreign Reference	JPH0877468-PDF.PDF	753639 8c7af5c8bc87c531d5920333f1968dd6c1519160	no	11
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8	Foreign Reference	JPH10162283-PDF.PDF	1008899 21f5461a653457cc0cc84966600a4cb588cb28f6	no	15
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9	Foreign Reference	WO2006054698-PDF.PDF	5448882 28699a5c6fd19433ee80f3e5f9470b2aea48f7cb	no	148
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Information:					
10	Foreign Reference	WO2009023784-PDF.PDF	2445774 fcbe09b3b2fe64e4cae6738bc064fc7f8be833a6	no	37
Warnings:					
Information:					
11	Foreign Reference	WO2012097150A1-PDF.pdf	2797231 f7d2f88516c4c50354ef4dbced8bf4b4ab52aaf5	no	32
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12	Non Patent Literature	IPR12.pdf	2364619 026370c6a3bda4105abd078fde3bbbdd396f6869	no	84
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13	Non Patent Literature	IPR13.pdf	2901733 c842c0d7b01864698898d4b3756b7f9e978d7b72	no	90
Warnings:					
Information:					
14	Non Patent Literature	IPR14.pdf	2715941 065ece6cb375d288dc25cf5ee67a640718c22002	no	83
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15	Non Patent Literature	IPR15.pdf	4597743 6dcf8ea05c77496ef2a891cb10131d18e1cd447d	no	95
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			Total Files Size (in bytes):	29519342	

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/736,180	CONFIRMATION NO. 5894
APPLICATION OF	:	STEVEN GOLDSTEIN	
DATE FILED	:	4 MAY 2022	
EXAMINER	:	UNKNOWN	
TC/A.U.	:	2654	
DOCKET	:	CONT-17203731	
TITLE	:	PERSONALIZED SOUND MANAGEMENT AND METHOD	

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: July 1, 2022

Sir:

The attached extensive IDSs contain references filed in recent IPRs, the IPRs themselves and similar references filed in invalidity contentions filed in Court for similar inventive scope. Applicant has compiled the extensive list for the Examiner's consideration and apologizes ahead of time if any of the references are copies of previously considered related art.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,
TECHIYA LLC

Date: July 1, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
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(Not for submission under 37 CFR 1.99)

Application Number	17736180
Filing Date	2022-05-04
First Named Inventor	Steven Goldstein
Art Unit	2654
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
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	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	10	3876843		1975-04-08	Moen	
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	13	8493204		2013-07-23	Wong et al.	
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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
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	26	8750295		2014-06-10	Liron	
	27	7920557		2011-04-05	Moote	
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	29	8611560		2013-12-17	Goldstein; Steven Wayne	
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	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	32	9053697		2015-06-09	Park	
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	39	9628896		2017-04-18	Ichimura	
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	41	6056698		2000-05-02	Iseberg	

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	43	8162846		2012-04-24	Epley	
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	46	6028514		2000-02-22	Lemelson	
	47	6408272		2002-06-18	White	

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	1	20030165246		2003-09-04	Kvaloy et al.	
	2	20050288057		2005-12-29	Lai et al.	
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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	Art Unit	2654
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	43	20070086600		2007-04-19	Boesen	
	44	20060287014		2006-12-21	Matsuura	
	45	20100119077		2010-05-13	Platz	

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	Application Number	17736180
	Filing Date	2022-05-04
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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
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Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

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☒ A certification statement is not submitted herewith.

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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-07-01
Name/Print	John P. Keady	Registration Number	56389

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(Not for submission under 37 CFR 1.99)

Application Number	17736180
Filing Date	2022-05-04
First Named Inventor	Steven Goldstein
Art Unit	2654
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	8045840		2011-10-25	Murata	

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	1	2006054698	WO		2006-05-26	Murata		☒
	2	2009023784	WO		2009-02-19	Usher		☐
	3	2012097150	WO		2012-07-19	Usher		☐

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

	4	H0877468	JP		1996-03-22	Keizaburo		☒
	5	H10162283	JP		1998-06-19	Masahiro		☒
	6	EP1519625	EP	A2		Victorian		☐
	7	2006037156	WO	A1		Mejia et al.		☐
	8	EP1401240	EP			Jensen		☐
	9	2008050583	WO			Kitada		☐
	10	2007092660	WO			Moberly		☐
	11	2004114722	WO			Pedersen		☐
	12	EP1385324	EP			Pedersen		☐
	13	EP1640972	EP			Dijkstra		☐

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	Art Unit	2654
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	Attorney Docket Number	CONT-17203731

Examiner Initials*	Cite No	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc), date, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]
	1	Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan 9-12, 2005, p. 284-286.	
	2	Bernard Widrow, John R. Glover Jr., John M. McCool, John Kaunitz, Charles S. Williams, Robert H. Hearn, James R. Zeidler, Eugene Dong Jr, and Robert C. Goodlin, Adaptive Noise Cancelling: Principles and Applications, PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975	
	3	Mauro Dentino, John M. McCool, and Bernard Widrow, Adaptive Filtering in the Frequency Domain, PROCEEDINGS OF THE IEEE, VOL. 66, NO. 12, DECEMBER 1978	

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[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

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OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-07-01
Name/Print	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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Application Number	17736180
Filing Date	2022-05-04
First Named Inventor	Steven Goldstein
Art Unit	2654
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	
	17	8140325		2012-03-20	Kanevsky	
	18	8983081		2015-03-17	Bayley	
	19	5327506		1994-07-05	Stites	

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	20	5933510		1999-08-03	Bryant	
	21	8086093		2011-12-27	Stuckman	
	22	7039585		2006-05-02	Wilmot	
	23	6606598		2003-08-12	Holthouse	
	24	6005525		1999-12-21	Kivela	
	25	RE38351		2003-12-16	Iseberg	
	26	7903825		2011-03-08	Melanson	
	27	8014553		2011-09-06	Radivojevic	
	28	8160261		2012-04-17	Schulein	
	29	6639987		2003-10-28	McIntosh	
	30	5923624		1999-07-13	Groeger	

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	31	8160273		2012-04-17	Visser	
	32	7464029		2008-12-09	Visser	
	33	7983907		2011-07-19	Visser	
	34	8254591		2012-08-28	Goldstein	
	35	8718305		2014-05-06	Usher	
	36	8798278		2014-08-05	Isabelle	
	37	8917894		2014-12-23	Goldstein	
	38	8625818		2014-01-07	Stultz	
	39	9491542		2016-11-08	Usher	
	40	8218784		2012-07-10	Schulein	
	41	6118877		2000-09-12	Lindemann	

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	Attorney Docket Number	CONT-17203731

	42	7050592		2006-05-23	Iseberg	
	43	8401200		2013-03-19	Tiscareno	
	44	9196247		2015-11-24	Harada	
	45	7574917		2009-08-18	Von Dach	
	46	5276740		1994-01-04	Inanaga et al.	

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	1	20030165246		2003-09-04	Kvaloy et al.	
	2	20080037801		2008-02-14	Alves et al.	
	3	20060204014		2006-09-14	Isenberg et al.	
	4	20050078838		2005-04-14	Simon	

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	5	20070036377		2007-02-15	Stirnemann	
	6	20020165719		2002-11-07	Wang	
	7	20060140425		2006-06-29	Berg	
	8	20030152359		2003-08-14	Kim	
	9	20070003090		2007-01-04	Anderson	
	10	20030165319		2003-09-04	Barber	
	11	20070255435		2007-11-01	Cohen	
	12	20040264938		2004-12-30	Felder	
	13	20060264176		2006-11-23	Hong	
	14	20040109579		2004-07-10	Izuchi	
	15	20060167687		2006-07-27	Kates	

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	Attorney Docket Number	CONT-17203731

	16	20060083390		2006-04-20	Kaderavek	
	17	20050068171		2005-03-31	Kelliher	
	18	20050028212		2005-02-03	Laronne	
	19	20080145032		2008-06-19	Lindroos	
	20	20020098878		2002-07-25	Mooney	
	21	20060188075		2006-08-24	Peterson	
	22	20140023203		2014-01-23	Rotschild	
	23	20060188105		2006-08-24	Baskerville	
	24	20020076057		2002-06-20	Voix	
	25	20110116643		2011-05-19	Tiscareno	
	26	20150215701		2015-07-30	Usher	

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	27	20140163976		2014-06-12	Park		
	28	20080137873		2008-06-12	Goldstein		
	29	20090122996		2009-05-14	Klein		
	30	20040086138		2004-05-06	Kuth		
	31	20070021958		2008-01-25	Visser et al.		
	32	20020111798		2002-08-15	Huang		

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CERTIFICATION STATEMENT

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See attached certification statement.

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SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-07-01
Name/Print	John P. Keady	Registration Number	56389

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8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17736180
Filing Date	2022-05-04
First Named Inventor	Steven Goldstein
Art Unit	2654
Examiner Name	
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
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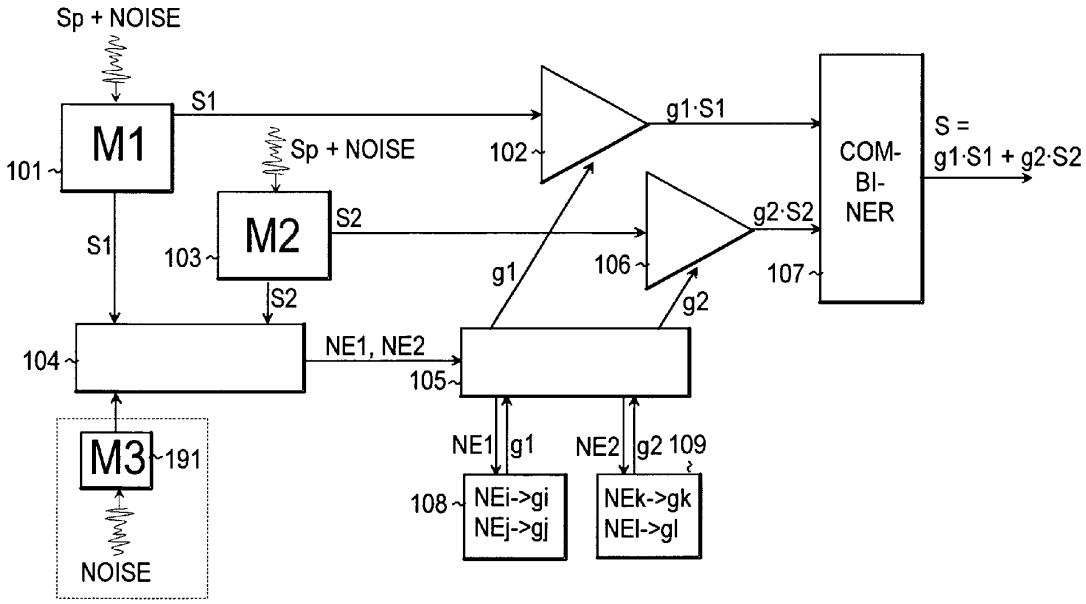
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(54) A system and method for reducing the effect of background noise

(57) Acoustical signals are collected with two microphones having different sensitivity to background noise. The level of background noise is estimated, and depending of the level of background noise, i) the amplification gain of the signals is modified before the signals are combined to a combined signal, or ii) the way of combining the signals is changed, to produce a signal that has less background noise than one of the microphones.

FIG 1



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Description

Field of the invention

[0001] The invention relates generally to enhancing speech quality in a telecommunications system, and more specifically to reducing the effect of background noise in the communication between users.

Background of the invention

[0002] Due to the recent explosive increase in the mobile device penetration, it is no more surprising to see people going on the sidewalk or pedestrian area with a mobile phone on their ear. The mobile phone is considered as a practical tool and it is supposed to be carried with to almost anywhere. People are supplied with means for being within reach in principle independently of time and location. As a consequence it may sometimes happen that the surroundings are not optimal for convenient telephoning. In fact, the sight of people trying to make a phone call in noisy places, such as restaurants or street corners, is getting more common all the time.

[0003] Technically, to provide the user with a microphone system which operates in a difficult acoustic environment has not been a difficult task in many years. For example, U.S. patent 5,327,506 discloses a speaker/microphone system that should be applicable in background noise levels up to far beyond over 100 decibels. Such a system is suitable for persons in high ambient noise environments, such as for helicopter pilots. The idea in 506 is that the outgoing voice sounds are taken from the air within the external auditory canal of a person while substantially eliminating audio vibration pickup of sound transmitted by bone conduction.

[0004] Another approach for high background noise microphone arrangements is found in U.S. patent 5,659,620. Here the microphone holder is designed for arrangement in its entirety in the outer ear outside the auditory meatus, and the microphone unit being mounted in vibration transmitting tissue contact in a part of the edge of the holder facing the bottom of tragus. As one of the benefits, all surround tones are not cut off thus increasing the comfort of wearing such a device. When people talk, vibrations in the throat and vocal cords are generated. The vibrations propagate through the bony and liquid-filled tissue, and the idea in 620 is that the jawbone which passes near the front part of the outer ear produces a strong transmission and transfers considerably more of the persons speech than if the signal were collected in the throat.

[0005] The idea of getting almost a background free aural image from inside the ear has also been adapted within the area of mobile communications, as illustrated in the international patent application WO 02/07477. In order to reduce and isolate bone conduction of sound, a sound isolation cup serves as a jacket that surrounds a miniature transducer inside the housing of a microphone/loudspeaker element that is placed in the ear.

[0006] However, under low background noise circumstances, the solutions above suffer from a poorer acoustical quality than an ordinary microphone system. In the 620 approach discussed above, the low spectrum dominates the transmitted sound spectrum because of the acoustical properties of the tissue. This means that in order to keep the sound quality acceptable, or at least to have a proper balance in the sound, the low frequencies have to be filtered. In the other two approaches discussed above, because the acoustic conduct from the speech producing organs to the ear is far from optimal, the sound quality gets badly distorted. Hence, prior art solutions can only be considered as enabling solutions, and in order to keep in pace with the challenging market, the wishes of the quality-conscious consumers of a mobile device with a good sound quality and a possibility to use the device also under high background noise conditions need to be fulfilled. The prior art solutions do not serve this purpose.

[0007] It is an object of the invention to bring about a system and method which provides the users of mobile devices with the best possible audio quality under high background noise circumstances and otherwise reduces the effect of the background noise.

Summary of the invention

[0008] This and other objectives of the invention are accomplished in accordance with the principles of the present invention by providing a system in which the processing of collected audio information is changed according to the estimated level of background noise thus reducing its overall effect.

[0009] More specifically, audio information is collected with two transducers having different sensitivity to the background noise. The background noise is estimated and i) amplification gain of the signals is modified before the signals are combined to a combined signal, or ii) the way of combining the signals is changed, which can be done either digitally or analogously.

[0010] According to the first embodiment of the invention, gain of at least one of the amplifiers adapted to amplify the detected signal is adjusted based on the noise estimate. The signal is then amplified using the adjusted gain thus producing at least one amplified signal. After this step, a combined signal is constructed from one or more of the amplified signals, and in the combined signal, the proportion of background noise is less than in one of the detected signals.

[0011] According to one aspect of the present invention, the system has also storage means for storing at least one noise estimate, and, correspondingly, also comparison means to compare said stored noise estimate with a predefined threshold value. After the comparison, a differential noise estimate is passed to gain control means.

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[0012] According to another aspect of the present invention, the system comprises storage means for storing at least two gain control parameters and comparison means for comparing at least two stored gain control parameters. The result of the comparison may be a differential gain parameter to be passed to the amplification means.

[0013] According to another aspect of the present invention, the gain control is adjusted to change the gain between the first and second microphones so that the gain of the first microphone is increased whereas the gain of the second microphone is decreased if the noise estimate indicates that the background noise level is above a first threshold value. Further, the gain control can be adjusted to change the gain of the second microphone substantially close to zero value if the background noise estimate shows that the background noise level is above a second threshold value. Respectively, they can be adapted to adjust the relative gain of the first microphone substantially close to one if the background noise estimate shows that the background noise level is above a pre-set threshold value.

[0014] In one preferred embodiment of the invention the first microphone is to be placed in the auditory canal of a user of the mobile handset; also a jawbone microphone can be used. The second microphone may be a microphone mounted in the mobile handset or in a vehicle, or it may be fixed in a hands-free set of a mobile handset. Examples of such mobile handsets where the invention may be used are mobile phones, PDA devices, laptop computers, dictation machines and so forth.

Brief Description of the Drawings

[0015] In the following, the invention and its preferred embodiments are described more closely referring to the examples shown in FIG. **1-7** in the appended drawings, wherein:

FIG. **1** illustrates one system for reducing the effect of background noise,

FIG. **2** illustrates another system for reducing the effect of background noise,

FIG. **3** shows one microphone set-up according to a preferred embodiment of the invention,

FIG. **4A** shows another microphone set-up according to another preferred embodiment of the invention,

FIG. **4B** illustrates a variation of the set-up shown in FIG. **4A**,

FIG. **5** shows one audio data collection set-up including two microphones,

FIG. **6** illustrates the relationship between noise level estimate and gains with which the signals detected by the microphones are amplified, and

FIG. **7** illustrates how a switch can be included into the system.

Detailed description of the invention

[0016] FIG. **1** illustrates a first system for reducing the effect of background noise, for example, in a microphone. Mechanical vibrations or pressure variations, preferably originating from speech $\mathbf{S_p}$ distorted by noise, is collected with a first transducer serving as a microphone **101**, producing a signal $\mathbf{S_1}$ which is then amplified with a gain $\mathbf{g_1}$ using an amplifier **102** to produce an amplified signal $\mathbf{g_1S_1}$. The system also includes another microphone **103** and a corresponding amplifier **106**, the microphone **103** similarly being exposed to speech $\mathbf{S_p}$ and background noise and producing first a signal $\mathbf{S_2}$ which is after amplification denoted by $\mathbf{g_2S_2}$, where $\mathbf{g_2}$ corresponds to the gain of the amplifier **106**. Both microphones **101** and **103** are transducers that convert acoustical or mechanical vibrations to an electric signal, i.e. normally to a small current, and they exhibit different sensitivity to background noise. The reason for the different noise sensitivity may be that the first microphone **101** is a bone conductor microphone or an earplug type microphone, whereas the second microphone **103** is a normal type transducer commonly used as a microphone in mobile terminals, and therefore more sensitive to noise than the aforementioned types of microphones.

[0017] Preferably, the microphones **101** and **103** are assembled to receive human speech $\mathbf{S_p}$, and in particular such a speech that is going to be further processed in a mobile terminal to be sent to a remote party of the discussion. It is by no means necessary that the speech $\mathbf{S_p}$ to be collected by the microphone (**101** or **103**) is originating from a single person, but may be originating from a conference call etc.

[0018] The output signals $\mathbf{S_1}$ and $\mathbf{S_2}$ of the microphones **101** and **103** are transmitted to a noise estimator **104** which includes means adapted to estimate background noise level thus producing noise estimates $\mathbf{NE_1}$ and $\mathbf{NE_2}$. The noise level can be estimated for both microphones **101** and **103** separately, the output signals **S1** and **S2** of both microphones **101** and **103** may be used to obtain a combined signal from which the noise is estimated, or only the output of the microphone **103** more sensitive to background noise may be used for noise estimation. In the latter two cases only one noise estimate is produced.

[0019] For example, the noise estimation can be done by averaging short periods of the signal **S1** or **S2**. During a silent period, i.e. when the user does not speak, the measured signal level is a good approximation of the

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background noise. This kind of approach is commonly used in the Discontinuous Transmission DTX widely used in the Global System for Mobile communications GSM, when instead of the coded silence only so called DTX parameters describing the nature of the background noise are sent, thus reducing obsolete usage of the air interface. The DTX parameters are received in the remote end or in the network between persons talking to each other and some noise similar to background is generated, calling this generated noise Comfort Noise. The idea of Comfort Noise comes from the observation that if during a telephone conversation the loudspeaker gets silent while one person is talking, he or she does not feel comfortable. Modern GSM mobile phones, for example, thus already have means for estimating and describing background noise. Similar means can be applied in the noise estimator 104 for serving the implementation of the invention.

[0020] The noise estimates $\mathbf{NE_1}$ and $\mathbf{NE_2}$ generated are passed to a gain control means **105**. The gain control means **105** adjust the gain $\mathbf{g_1}$, $\mathbf{g_2}$ of the both amplifiers **102** and **106**. In the adjustment, a noise estimate $\mathbf{NE_1}$ or $\mathbf{NE_2}$ is used to determine the desired amplification gain. For this purpose the system may include storage means for noise-amplification relationships, including means **108** and **109**, e.g. tables or functions which respond with a desired gain level $\mathbf{g_x}$ to a query including a corresponding noise level $\mathbf{NE_x}$. For example, means 108 return a gain value $\mathbf{g_1}$ in response to noise estimate $\mathbf{NE_1}$, and means **109** return a gain value $\mathbf{g_2}$ in response to noise estimate $\mathbf{NE_2}$.

[0021] In other words, means **108** and **109** tell the gain control means **105** what is the optimal relation between a noise estimate NE_x and an amplifier gain g_x. Means **108** and **109** are preferably microphone specific, i.e. there are two tables in the system. The gain control means **105** get the noise estimate NE_x, retrieve the gain values g_x corresponding the estimate of present noise NE_x and determine how the gain $\mathbf{g_1}$ and $\mathbf{g_2}$ of the amplifiers **102** and **106** is to be adjusted. In principle, the gain values $\mathbf{g_1}$ and $\mathbf{g_2}$ get values between 0 and 1 so that $g_1 + g_2 = 1$.

[0022] After passing the gain adjustment information $\mathbf{g_1}$ and $\mathbf{g_2}$ to the amplifiers **102** and **106**, the amplifiers amplify the signals S_1 and S_2 using the amplifier gain set based on the gain adjustment information. After amplifying, a combined signal **S** is constructed in the combining means **107**, the construction being made using the amplified signals $\mathbf{g_1S_1}$ and $\mathbf{g_2S_2}$. This may include changing the delay and phase of at least one of the signals g_xS_x so that the distortion resulting from the amplification and the signal path is minimised, and finally summing up the signals g_xS_x so that the combined signal **S** represents a linear combination of the signals $\mathbf{S_1}$ and $\mathbf{S_2}$ detected by the microphones **101** and **103**. This constructed signal S is then ready for being processed further, including all of the numerous steps necessary for transmitting the audio information over the air interface. The combined signal **S** has a smaller proportion of background noise than the output signal $\mathbf{S_2}$ of microphone **103** collected. In this way the effect of background noise can be reduced.

[0023] Preferably, the tables **108** and **109** are generated so that corresponding to a change ΔNE ($=NE_{t=t2}-NE_{t=t1}$) in the noise level NE_x there is a corresponding change Δg ($=g_{t=t2}-g_{t=t1}$) in the gain value g_x. As characteristic to one aspect of the present invention, the gain adjustments of the amplifiers **102**, **106** are opposed to each other; i.e. when the background noise is increasing, the gain $\mathbf{g_1}$ of the signal detected by the first (low-noise) microphone **101** is increased whereas the gain $\mathbf{g_2}$ of the signal detected by the second (more sensitive to background noise) microphone **103** is decreased. It is favourable to normalise the gains so that the relative gain for the both microphones equals to 1.

[0024] In other words, the means **108** and **109** are modified to correspond to the different noise behaviour of the two transducers so that the linear combination of the signals $\mathbf{S = g_1S_1 + g_2S_2}$ detected by the two transducers serving as microphones **101** and **103** can be subjectively (by listening) considered optimal. This may include different threshold values: The first threshold value corresponds to the value of at least one of the noise estimates, at which value the quality of signal $\mathbf{S_2}$ of the second microphone **103** that is more sensitive to background noise starts to deteriorate; this is the point where the relative gain of the first microphone **101** is increased from substantially zero gain value to a small value. The second threshold value corresponds to the value of at least one of the noise estimates NE_x at which the quality of the detected signal of the second microphone **103** is so much worse than the quality of the signal $\mathbf{S_1}$ detected by the first microphone **101** that the relative gain $\mathbf{g_2}$ of the second microphone **103** is set to substantially zero gain.

[0025] As an example, if the first microphone detects a background noise level of 2,0 mV (corresponding to a noise 70 dB) and the second microphone detects a background noise level of 5,0 mV (corresponding to a noise of 72 dB), the average noise level estimated in the noise estimator may be between 70 and 72 dB. The relative gains for the microphones read from the table are then 0,4 and 0,6, respectively. This is not straightforward, because the sensitivity of the microphone plays a significant role; the signal levels may need to be adjusted to correspond some reference level. The relative gains are then to be understood against this reference level only.

[0026] When the detected background noise jumps to 2,2 mV (71 dB) and 7,0 mV (76 dB), the average noise level may be estimated to be between 71 and 76 dB. The relative gains for the microphones read from the table are then 0,95 and 0,05, respectively. The second microphone is more sensitive to the background noise, i. e. the background noise generates a bigger impulse in the microphone. Secondly, the second microphone is

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not yet shut-off completely, because the noise level is not above a defined shut-off threshold value for background noise.

[0027] The transition at different moments of time between subsequent values of the relative signal $\mathbf{S_1}$ and $\mathbf{S_2}$ proportions in the combined signal **S** defined by a ratio $\mathbf{g_1/g_2}$ (when $g2 \neq 0$) can thus be made smooth or at least to correspond small steps in order to avoid abrupt transitions that might be considered annoying by the listener. This corresponds to successive changes of $\mathbf{g_1}$ and $\mathbf{g_2}$. Nevertheless, it is to be noted that this is not necessary, but the transition may resemble a step function, i.e. the gain $\mathbf{g_1}$ for the signal detected by the first microphone **101** is zero below a threshold value and one otherwise, and the gain $\mathbf{g_2}$ for the signal detected by the second microphone is then reversed, i.e. it obtains values $1\text{-}\mathbf{g_1}$.

[0028] FIG. **6** shows an example of the relation between estimated noise level $\mathbf{NE_2}$ estimated from the signal $\mathbf{S_2}$, and the amplifier gain g_1. In this case, $g_2=1\text{-}\mathbf{g_1}$. At times preceding **TA**, $\mathbf{NE_2}$ is smaller than the first threshold value. Correspondingly, $\mathbf{g_1}=0$ and $g_2=1$. At **TA**, the value of $\mathbf{NE_2}$ is increasing, so $\mathbf{g_1}$ is stepwise increased. At **TB**, $\mathbf{NE_2}$ has reached the second threshold value, and $\mathbf{g_1}$ is set to one. At **TC**, $\mathbf{NE_2}$ is again smaller than the second threshold value, so the value of **g1** must be decreased. At **TD**, $\mathbf{NE_2}$ falls below the first threshold value and **g1** is set to zero value. At the time interval $\mathbf{P_1}$ the change in $\mathbf{NE_2}$ is less than the threshold $\Delta\mathbf{NE}$ set for the change of **NE**, so during this interval $\mathbf{g_1}$ is not changed. Immediately after $\mathbf{P_1}$, when the change in $\mathbf{NE_2}$ is larger than the threshold value, the $\mathbf{g_1}$ is changed by $\Delta\mathbf{g}$. During the interval $\mathbf{P_2}$, $\mathbf{g_1}$ is not changed because $\mathbf{NE_2}$ shows sharp peaks; this may present some distortion so the changes are neglected.

[0029] The second predefined threshold value can be chosen to be a fixed value as well. The value may be experimentally determined to correspond to the level of background noise in which the input signal of the second microphone **103** has been deteriorated in such a bad way that in order to get a better signal quality for the signal **S**, only the input signal $\mathbf{g_1S_1}$ is used.

[0030] FIG 7 shows how a second system according to the invention, the system having only a stepwise transition for the value of $\mathbf{g_1}$. Speech **Sp** and noise are detected by microphones **101** and **103** producing signals **S1** and **S2**. Background noise is collected by a third microphone **191** directly connected to noise estimator **104**. The noise estimator estimates the level **NE** of background noise and passes it to gain control means **705**. The gain control means **705** are adapted to compare the noise level estimate **NE** with a predefined threshold value. If the value of **NE** is less than the threshold value, **g1** is set to zero, otherwise **g1** gets the value of 1. The value of **g1** indicates the position of a switch **SW1** controlled by the gain control means **705**, the switch **SW1** being so arranged that the terminals **TE1** and **TE2** are connected when **g1=1**; **TE1** and **TE3** are connected in all other cases. The terminal **TE1** leads to amplification means **702** which produce an amplified signal **AS**. Because only one signal is fed (S1, if TE1 and TE2 are connected) to the amplifier, no literal combining of the signals (S1, S2) need to be performed.

[0031] The switch turns the relative gain $\mathbf{g_2}$ of the second microphone to zero and the relative gain of the first microphone $\mathbf{g_1}$ to one. In this case the function of the switch **SW1** and amplification means **702** may be understood as linear amplification with amplification factor 1.0 when the signal received by a microphone is amplified and zero otherwise.

[0032] Further, in order to avoid annoying abrupt transitions between the first and second microphones, or to avoid fluctuations in the microphone gains, the system like presented in FIG. **1** may comprise also storing means adapted for storing noise parameters in order to form a noise history. The noise parameters can be calculated for one microphone, both microphones separately or they can be obtained by combining the values computed from the inputs of the both microphones. One possibility is to use the second microphone input as the preferable source for noise estimation, as the first microphone is less sensitive to background noise.

[0033] If history data is collected, the system may further be adapted to have a hysteresis effect, i.e. if the noise level is having fluctuations back and forth, the system can adapt the threshold value to correspond a larger shift in the background noise. In order to have a hysteresis effect, there are storage means and comparison means; the hysteresis information may be obtained from a stored table, the table being read in a way taking into account the noise level history.

[0034] Similarly, the history data may be used for making prognoses or extrapolations for the noise level, and in this way the system may, based on the noise level measured at least in two earlier moments of time, predict a noise level in the future thus making the possibly larger transitions in gains over a longer period of time.

[0035] A third microphone **191** is optionally included in the system. Its main purpose is to detect background noise. It may be located in the vicinity of the noise estimation means **104** so that the transmitting of the signal to the noise estimation means **104** from the microphone **191** is straightforward. In this case the signals $\mathbf{S_1}$ and/or $\mathbf{S_2}$ need not to be passed from the other two microphones **101** and **103** to the noise estimator **104**. This makes the system simpler, as the signal from microphones **101** and **103** need only travel the path to amplifiers **102** and **106**. For example, in a wireless hands-free set the microphone **101** is located in one ear of the user, whereas the second microphone **103** is fixed in the car of the user. The amplifiers are located close to the microphones, especially because the signals are to be transmitted over the air interface to the combiner means **107**. This means that the noise estimator **104** and gain control means **105** need to instruct the amplifiers **102** and **106** separately; the noise may be measured by the

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microphone **191**, and the gain parameters $\mathbf{g_1}$ and $\mathbf{g_2}$ be broadcast to the amplifiers **102** and **106**.

[0036] FIG. **2** shows a third system according to the present invention. The transducers **101** and **103** are exposed to speech signal $\mathbf{S_p}$ distorted by background noise. They detect acoustical vibrations, i.e. mechanical vibrations or pressure changes. Amplifiers **202** and **206** amplify the signals $\mathbf{S_1}$ and $\mathbf{S_2}$ with constant gains thus producing amplified signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$, respectively. An Analog to Digital Converter ADC **207** converts the signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$ pre-amplified by amplifiers **102** and **106** into a digital form. In addition to performing this step, the **ADC** may further include combining means to form a combined signal S from at least one of the amplified signals $\mathbf{AS_1}$ and $\mathbf{AS_2}$ in such a manner that the relative ratio of the signals may be altered. The combining may be performed digitally, thus possibly including multiplication and summing operations, or it may be performed for the analog signals before the conversion.

[0037] The noise level is detected in noise estimator **204**, including tables **208** and **209** showing noise responses of the microphones **101** and **103** in a manner similar to the tables discussed above. The noise estimates NE_x are then used to deduce the target gain factors g_x. The target gain factors $\mathbf{g_1}$ and $\mathbf{g_2}$ which correspond to combining parameters are lead to the combining means of the ADC **207**, where the ADC either, based on the combining parameters g_1 and g_2, performs the conversion and only after that digitally multiplies the converted signal by the gain factor, or, uses the target gain factor to change the relative strength of the signals before the conversion.

[0038] In a fourth system according to the present invention, the system may have storage means for storing at least one noise estimate $\mathbf{NE_1}$ or $\mathbf{NE_2}$ in the noise estimator **104**, whereas the system otherwise corresponds to the system already discussed with reference to FIG. **1**. This means that at least one noise estimate $NE_t=_{tpast}$ value may be stored, and, correspondingly, at a later point in time $t=t_1$, at least one other noise level estimate $NE_{t=t1}$ may be stored as well. The stored noise estimates may be compared using comparison means located preferably in the noise estimator **104**. The comparison may involve computing a noise level difference $\Delta NE = NE_{t=t1}-NE_{t=tpast}$ describing the changing of the background noise level. This difference may be transmitted then to gain control means **105** which change gains according to a function $\Delta g=f(\Delta NE)$. It is also possible that the difference is not passed to gain control means **105** if it does not exceed a preset threshold value, i.e. if $\Delta NE < \Delta NE_{threshold}$; only changes in the gain when the change in the background noise level is larger than the threshold value are passed to the amplifier **102** or **106**. In this case the older value measured at t_{past} may remain stored in the register and newer noise estimates $NE_{t=t1'}$ calculated at times $t_1' > t_1$ are overwritten on the value $NE_{t=t1}$.

[0039] A fifth system in accordance with the present invention, the system comprises storage means such as a register in the gain control means **204** in addition to other components as discussed with reference to FIG. **2**. The storage meansmay be used for storing at least two gain control parameters. Further, the system may also include comparison means for comparing at least two stored gain control parameters. Like discussed above, the comparison of the parameters may include performing some mathematical operations, like calculating the difference between a newer and an older gain parameter value, and, in the case that a pre-set threshold value for the difference is exceeded, which can be checked again by comparison, the system may calculate a new gain to be passed to the ADC **207**. The new gain may also be selected from a plurality of different gain set-ups, if the system is to be kept simple and fast.

[0040] FIG. **3** shows a microphone set-up according to one preferred embodiment of the present invention. The mobile phone in the figure has two microphones **M1** and **M2**. The second microphone **M2** is an ordinary microphone denoted by asterisks **30** describing holes in the cover of the bottom half of the phone. The first microphone **M1** is located close to the loudspeaker which is denoted by asterisks, whereas the microphone **M1** is denoted by arc **31**. Because the loudspeaker is - usually - placed close to the ear, using a jawbone contact element as the microphone **M1**, the vibrations in the jawbone of the user can be detected thus producing a signal S_1. The sensitivity of **M1** to disturbances from the loudspeaker is reduced because **M1** detects mechanical vibrations from the jawbone of the user.

[0041] FIG. **4A** further illustrates a microphone set-up according to one other preferred embodiment of the present invention. The first microphone **M1** is located in an earplug-like element which can measure the air vibrations in the ear canal (i.e. auditory canal) of the user of the mobile device. Depending of the actual realisation of the invention, this earplug microphone can be similar to those disclosed in the referenced publications. Some of the prior art earplug microphones have a wireless transmitter, such as a BLUETOOTH transmitter.

[0042] A solution according to FIG. **4A** is especially suitable for different hands-free applications. For example, many countries in Europe have recently adopted restrictions for usage of mobile phones while driving a vehicle. In order to comply with the legislation, a driving mobile user willing to receive a phone call need either to stop or to have a hands-free equipment installed in his or her car.

[0043] Some hands-free equipment sets include a speaker, a microphone and then some wiring systems, possibly with an extra antenna. The speaker is typically mounted somewhere in the cabin, whereas the microphone is located close to the head of the driver. As known, even though some of the different speech codecs widely used in the mobile devices enhance the speech quality in noisy environments, such as in a moving car, the background noise may still be annoying and,

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depending of the car, even make the discussion impossible, depending on the circumstances such as other traffic, speed of the vehicle, road conditions etc. This drawback can be eliminated if the user is supplied with an earplug-like device in addition to the normal hands-free set installed in the car. As an alternative, a jawbone microphone can be used as well, while there are some of them available on the market already. Preferably the first microphone (either earplug or jawbone) is equipped with a wireless transmitter. Then either the mobile handset or the hands-free set includes a receiver to receive the transmitted audio information, such as BLUETOOTH low-power RF data.

[0044] The transmitter needs not to be wireless but may be a conventional earplug microphone as well like illustrated in FIG. **4B**, where there is a wire including two lines, one for the signal and one for the ground level. In the other end the wires may be connected to the front panel of the car, the panel supplying a plug-in for the microphone data and further connected to the hands-free set, or directly to the mobile handset.

[0045] FIG. 5 illustrates another option for implementing the invention. Here the first and second microphones are included in a headset kit of a mobile phone. Typically, the headset kit includes a plug **501** which is connected to a mobile phone or a car front panel. The handset provides hands-free operation and may also include an answer/end button located in the second microphone **M2** or somewhere else in the cable. The earplug including a loudspeaker may also include an earplug microphone **M1**, provided that the acoustical separation of the loudspeaker and microphone can be guaranteed, or a jawbone microphone, where the acoustical separation should be a minor problem.

[0046] For a person skilled in the art it is also clear that no implicit tables like those described with reference to FIG. **1** and **2** are necessary. Depending on the number of steps for different amplifications needed, the system can be coded directly if the number is low, otherwise also a numerical model which interpolates some experimentally tested values may also be used.

[0047] Although the invention was described above with reference to the examples shown in the appended drawings, it is obvious that the invention is not limited to these, but it may be modified by those skilled in the art without departing from the scope and the spirit of the invention.

Claims

1. A system for reducing the effect of background noise especially in a mobile handset, the system comprising

   - a first transducer (101) for detecting an acoustical signal producing a first signal detected (S1),
   - a second transducer (103) for detecting an acoustical signal producing a second signal detected (S2), the first and second transducers having different sensitivities to background noise,
   - amplifier means (102, 106) for amplifying the first and/or second signal (S1, S2) detected thus producing at least one amplified signal (g1S1, g2S2),
   - noise estimator means (104) adapted to estimate a background noise level present at least in one of the detected signals (S1, S2) or in a signal measured by a third transducer (191) thus producing at least one noise estimate (NE1, NE2),
   - gain control means (105), responsive to said noise estimator means (104, 204), adapted to adjust gain (g1, g2) of the amplification means (102, 106, 207), the adjustment being made depending on at least one noise estimate (NE1, NE2), and
   - combining means (107), adapted to form a combined signal (S) from at least one amplified signal (g1S1, g2S2) thus reducing the effect of background noise.

2. A system according to claim 1, **wherein** the noise estimator means are connected to the first and/or second transducer (101, 102) for receiving noise level information.

3. A system according to claim 1 or 2, the system further **comprising**

   - storage means for storing at least one noise estimate, and
   - comparison means for comparing the stored noise estimate with a predefined threshold value, the comparison producing a differential noise estimate to be passed to gain control means (105).

4. A system according to claim 1 or 2, the system further **comprising**

   - storage means for storing at least two gain control parameters, and
   - comparison means for comparing the two gain control parameters, the comparison producing a differential gain parameter to be passed to the amplification means.

5. A system according to claim 1 or 2, **wherein** the gain control means (105) are adapted to adjust gain (g1, g2) between the first and second amplifiers (102, 106) so that the gain (g1) of the first amplifier (102) is increased whereas the gain (g2) of the second amplifier (106) is decreased if the noise esti-

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mate (NE1, NE2) indicates that the background noise level is above a first threshold value.

6. A system according to claim 1, 2 or 5, **wherein** the gain control means (105, 207) are adapted to adjust gain (g2) of the second amplifier substantially close to zero value if the background noise estimate (NE) shows that the background noise level is above a second threshold value.

7. A system according to claim 1, 2, 5 or 6, **wherein** the gain control means (105, 207) are adapted to adjust the relative gain of the first transducer substantially close to one if the background noise estimate (NE) shows that the background noise level is above a pre-set threshold value.

8. A system according to claim 1, **wherein** the construction is performed by adding the amplified signals (g1S1, g2S2).

9. A system according to any one of the preceding claims, **wherein** the first transducer (101) is a microphone to be located in the auditory canal of a user of the mobile handset.

10. A system according to any one of the preceding claims, **wherein** the first transducer (101) is a jawbone microphone.

11. A system according to claim 9 or 10, **wherein** the second transducer (103) is a microphone mounted in the mobile handset.

12. A system according to claim 9 or 10, **wherein** the second transducer (103) is a microphone fixed in a hands-free set of a mobile handset.

13. A system according to claim 1 or 2, **wherein** said gain control means (105, 207) are located in the mobile handset or in the hands-free set of a mobile handset.

14. A method for reducing the effect of background noise especially in a mobile handset, the method **comprising** the steps of

- collecting a first acoustical signal (S1),
- collecting a second acoustical signal (S2), whereby in the collected signals (S1, S2) a different level of background noise is present,
- estimating background noise level from at least one of the signals (S1, S2) collected,
- in response to said estimating step, adjusting gain (g1, g2) of at least one amplifier (102, 106) adapted to amplify signals collected (S1, S2),
- in response to said adjustment step, amplifying said collected first and/or second signal (S1, S2) to produce at least one amplified signal (g1S1, G2S2), and
- in response to said amplifying step, constructing a combined signal (S) from at least one amplified signal (g1S1, g2S2) thus reducing the effect of background noise in the combined signal.

15. A method according to claim 14, further **comprising** the steps of

- producing and storing at least one noise estimate (NE1, NE2) in the background noise estimation step, and
- comparing said stored noise estimate (NE1, NE2) with a predefined threshold value, the comparison result being used in the gain adjustment step.

16. A method according to claim 14, further **comprising** the steps of

- in the gain adjusting step, storing at least two gain control parameters, and
- comparing at least two stored gain control parameters in order to decide whether the gain adjusting step is to be taken, the gain adjusting step being performed only if the comparison result shows that a gain control parameter being larger than a predefined threshold value.

17. A method according to claim 14, **wherein** the gain adjustment step is adapted to adjust amplification balance between the signals detected by first and second transducers (101, 103) so that the gain (g1) of the amplifier (101) amplifying the first signal (S1) collected is increased whereas the gain (g2) of the amplifier (106) amplifying the second signal (S2) is decreased if in the noise estimation step it is detected that the background noise level is higher than a first threshold value.

18. A method according to claim 14, **wherein** the gain adjustment step is adapted to adjust gain (g2) of the amplifier amplifying the second signal (S2) substantially close to zero value if the background noise estimate (NE) shows that the background noise level is above a second threshold value.

19. A method according to claim 14, **wherein** the construction step includes adding the amplified signals (g1S1, g2S2).

20. A method according to any one of claims 14-19, **wherein** the first signal is collected using a microphone to be located in the auditory canal of a user of the mobile handset.

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21. A method according to any one of claims 14-19, **wherein** the first signal is collected using a jawbone microphone.

22. A system for reducing the effect of background noise especially in a mobile handset, the system **comprising**

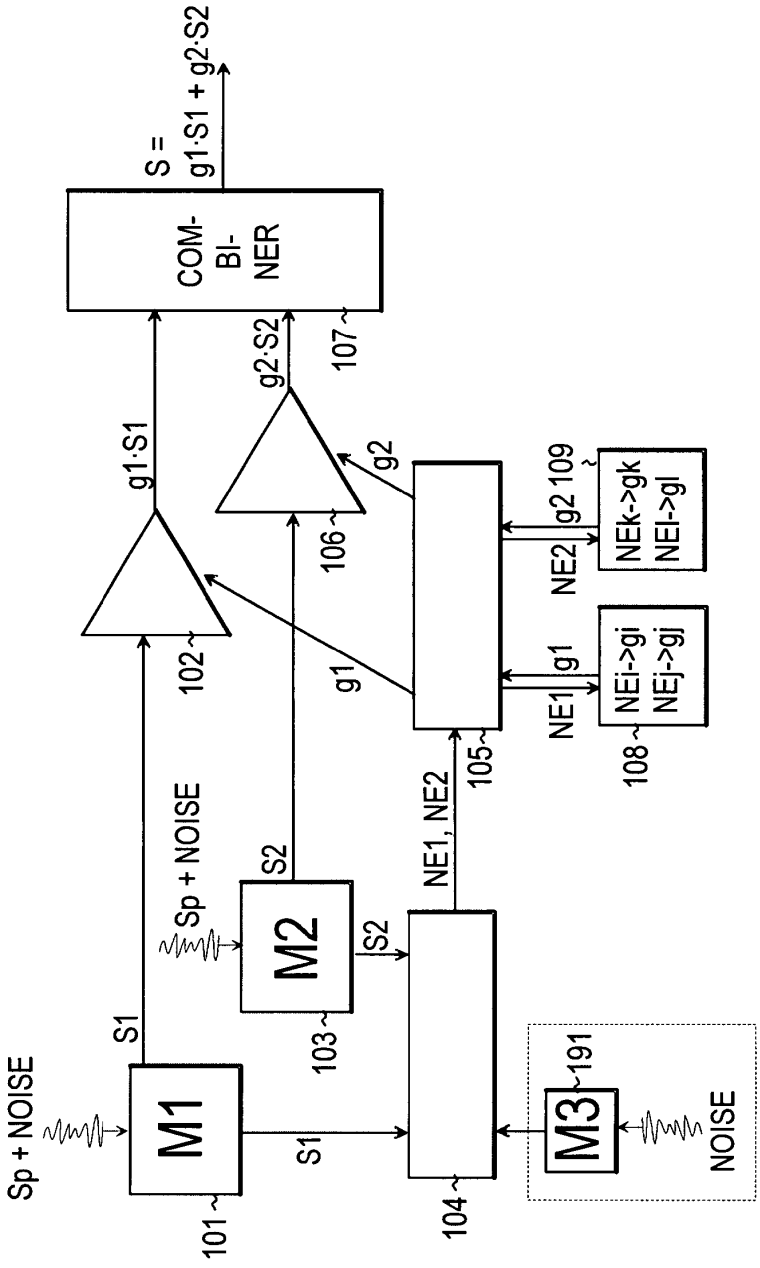
- a first transducer (101) for detecting an acoustical signal producing a first signal detected (S1),
- a second transducer (103) for detecting an acoustical signal producing a second signal detected (S2), the first and second transducer (101, 103) having different sensitivity to background noise,
- amplification means (202, 206) for amplifying the first and/or second signal (S1, S2) detected thus producing at least one amplified signal (AS1, AS2),
- combining means (207) adapted to form a combined signal (S) from at least one of the amplified signals (AS1, AS2) using at least one combining parameter (g1, g2) thus reducing the effect of background noise,
- noise estimator means (204) adapted to estimate background noise level present at least in one of the detected signals (S1, S2) or in a signal measured by a third transducer (191) thus producing at least one noise estimate (NE1, NE2),
- combining parameter generation means (208, 209) adapted to generate at least one combining parameter (g1, g2) in response to at least one noise estimate (NE1, NE2), and
- means (204) for passing the generated combining parameter (g1, g2) to the combining means (207).

23. A method for reducing the effect of background noise especially in a mobile handset, the method **comprising** the steps of

- collecting a first acoustical signal (S1),
- collecting a second acoustical signal (S2), whereby in the collected signals a different level of background noise is present,
- estimating background noise level (NE1, NE2) from at least one of the signals collected or from a third acoustical signal,
- in response to the background noise estimating step, using at least one background noise level parameter (NE1, NE2) for selecting at least one combining parameter (g1, g2), and
- in response to said selecting step, using said summing parameter for combining the first and acoustical signals (S1, S2) to form a combined signal (S) which has less background noise than the second signal collected.

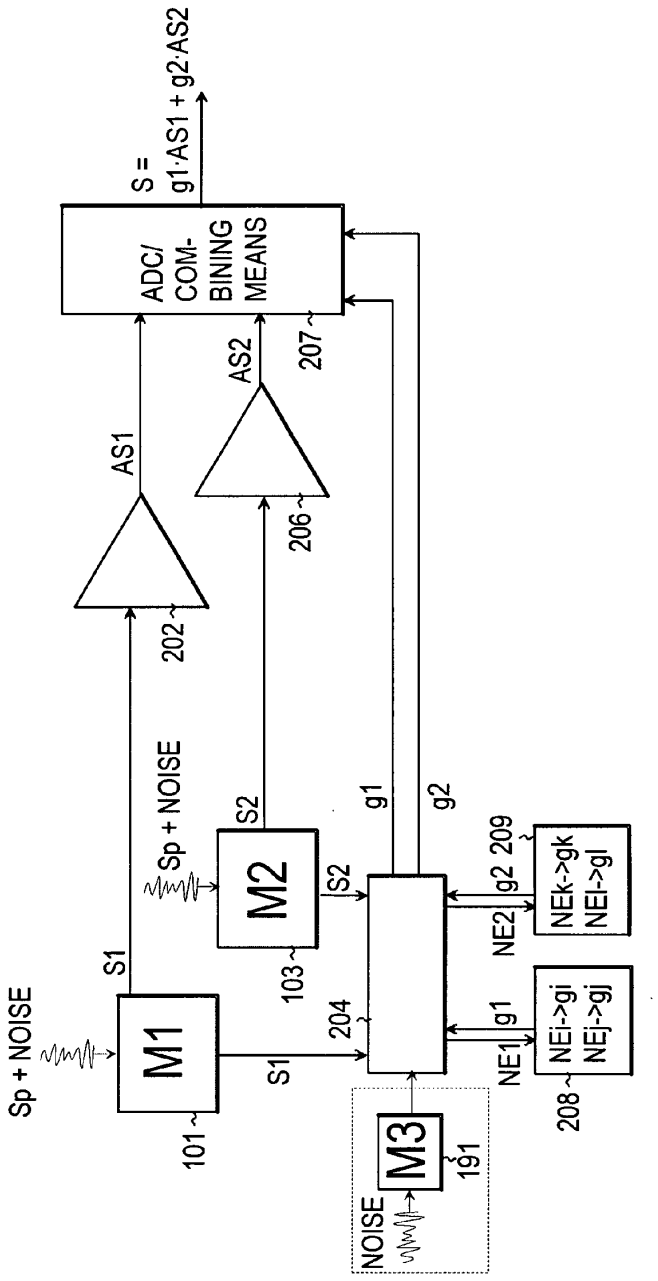
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FIG 1



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FIG 2


S = g1·AS1 + g2·AS2
ADC/ COM- BINING MEANS
207
AS2
206
AS1
202
Sp + NOISE
S2
M2
103
S1
Sp + NOISE
M1
101
S1
204
S2
g1
g2
NE2
g2
209
NEk->gk
NEl->gl
NE1
g1
NEi->gi
NEj->gj
208
NOISE
M3
191

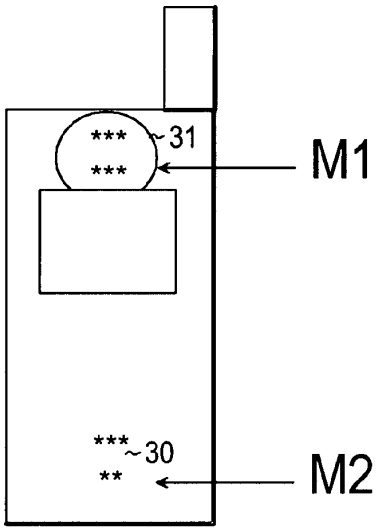


FIG 3
(LEFT)

FIG. 4A

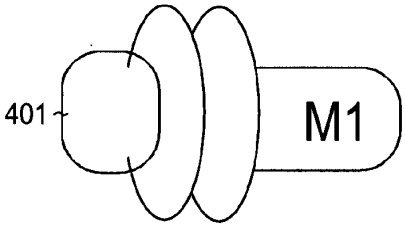
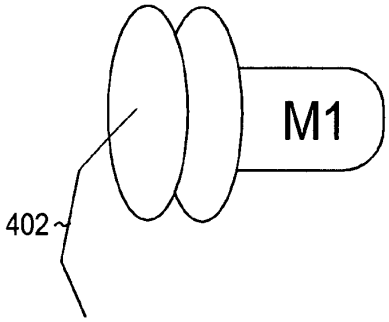


FIG 4B



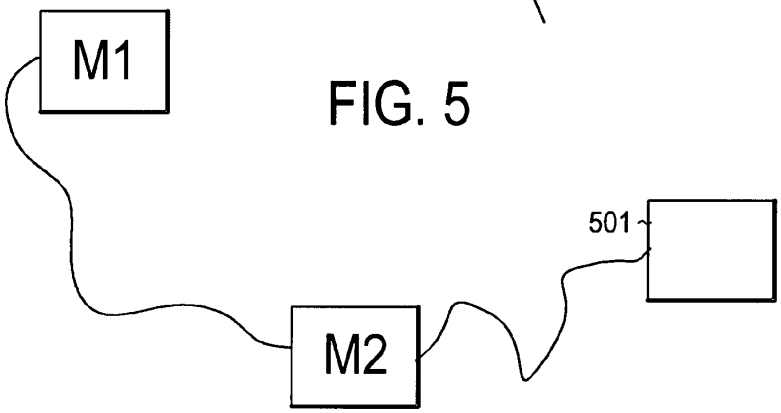


FIG. 5

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FIG 6

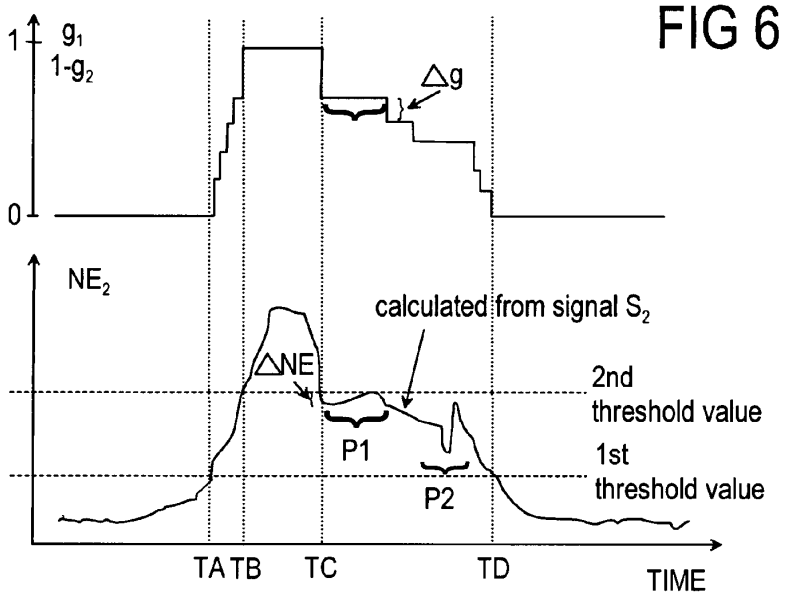
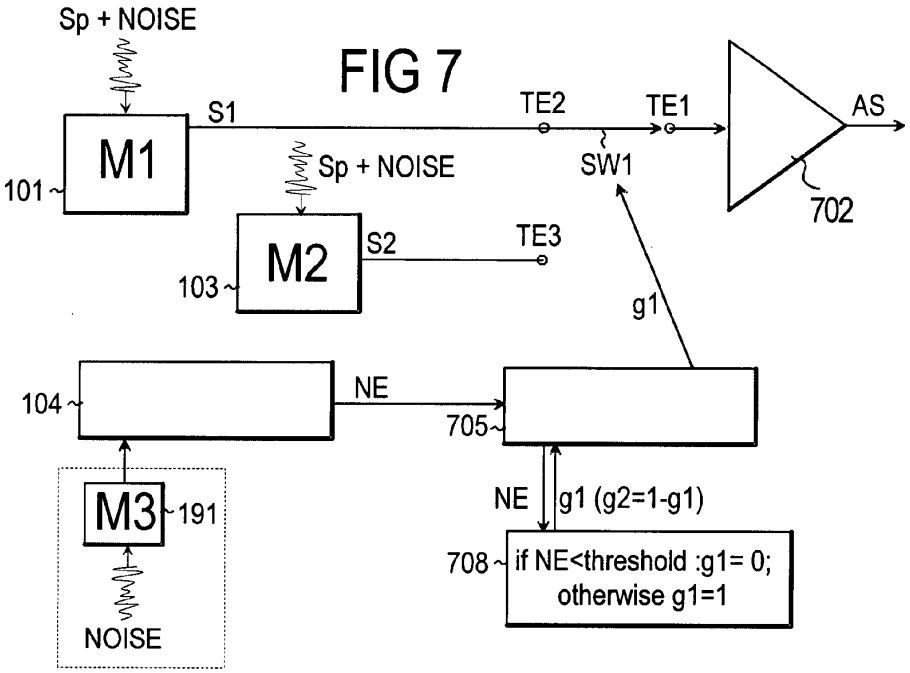


FIG 7



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European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 01 6435

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
A	US 5 640 450 A (WATANABE OSAMU) 17 June 1997 (1997-06-17) * abstract * ---	1-23	H04M9/08 H04M1/60
A	EP 0 767 570 A (NOKIA MOBILE PHONES LTD) 9 April 1997 (1997-04-09) * abstract * ---	1-23	
A	WO 99 05840 A (QUALCOMM INC) 4 February 1999 (1999-02-04) * abstract * -----	1-23	

TECHNICAL FIELDS SEARCHED (Int.Cl.7)

H04M

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	2 December 2003	Montalbano, F

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

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ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 02 01 6435

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report. The members are as contained in the European Patent Office EDP file on
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02-12-2003

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

(19) Europäisches Patentamt
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(11) EP 1 401 240 A1

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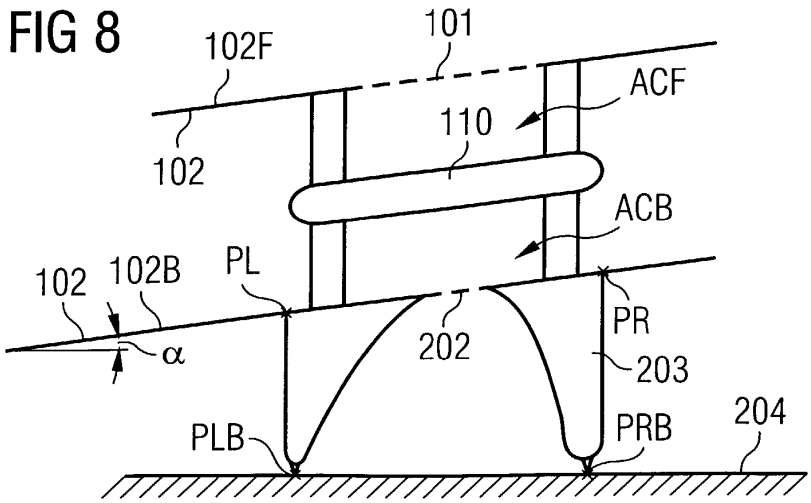
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(54) A dual directional mode mobile terminal and a method for manufacturing of the same

(57) A mobile terminal is equipped with an acoustoelectrical transducer arranged in a housing of the mobile terminal, the housing comprising i) a user sided first inlet aperture in communication with the acoustoelectrical transducer, ii) a second inlet aperture also in communication with the acoustoelectrical transducer, and iii) sealing means for sealing the second inlet aperture when the mobile terminal is placed on an external surface or object. As a result, the mobile terminal has selectable directional characteristics.

FIG 8



EP 1 401 240 A1

Printed by Jouve, 75001 PARIS (FR)

Description

Field of the invention

[0001] The invention relates generally to enhancing speech quality in a telecommunications system, and, more specifically, to enhancing directional properties of a microphone in a mobile terminal.

Background of the invention

[0002] The mobile device boom of the recent years has resulted in different new uses for the mobile terminals widely used in telecommunications. The users expect their new devices to be usable in many different situations. The expectations regarding to the acoustical quality of the device are high, and because the device may be used for various purposes, this sets very high requirements for the design of the terminals.

[0003] As an example of the dual use of the mobile handset, it is very common that the terminal, in addition to its normal use as a handset, is used as a hands-free device. This may be the case when the user is willing to use the mobile terminal in a vehicle, for example, or use it for conference calling. Typically, in the latter case, the mobile terminal is set on a table, the persons willing to speak then usually grouping around the terminal.

[0004] From the acoustical design point of view, the latter kind of use requires quite different acoustical and directional properties of the microphone than the first use mentioned. In conference calling, it is very critical that the microphone shows omni-directional characteristics so that the position of the person speaking does not influence too radically on the sound quality, and when one person is speaking, also comments from the background should be carefully detected.

[0005] On the other hand, when a mobile terminal is used as a handset, it is preferable that the background noise is not detected by the microphone. Normally, the mouth and other sound producing organs of the mobile user are within a predetermined distance from the microphone when he or she is speaking to his or her mobile handset. If human anatomy factors are taken into account, it is easier to design a phone in such a way that the mouth of an average male/female user is quite close to the centre of inlet of the microphone. Thus the microphone should preferably show unidirectional characteristics in order to receive the speech of the user but not noise from his or her background.

[0006] Figure 1 illustrates a prior art mobile terminal 100. The terminal 100 includes housing 102, wherein a microphone 110 is assembled. For the microphone 110 an aperture 101 is important so that the energy of acoustic signals is not attenuated too much which would be the case if the acoustic signals had to traverse through the housing 102.

[0007] Having both omni- and unidirectional characteristics in the acoustical parts of one mobile terminal is not a straightforward task. Some prior art solutions that might be applicable to at least some extent can be found, but they present drawbacks which both an ordinary user and an acoustics designer would find annoying.

[0008] International patent publication WO 01/69967 A2 shows one example of implementing a selectivity switch between a directional mode and an omni-directional mode in a hearing aid device. The switch includes a switch body having a port aperture therein in communication with one of the ports of the microphone, and a switch actuator moveably connected to the switch body and moveable to a first position wherein the port aperture is generally unobstructed by the actuator and a second position wherein the actuator covers the port aperture.

[0009] In the modern mobile lifestyle, a mobile terminal sometimes presents some values of the user of the terminal. Thus, the aesthetic design is very important, and the switch does not support the purpose of creating an aesthetically enjoyable device. Secondly, the shape and form of a hearing aid device is quite different to a mobile phone, basically because the hearing aid device is located in the ear of the user, so the mechanics of the device can be designed in a completely different manner. Therefore the '967 solution cannot be without problems used for ordinary mobile terminals.

[0010] Another solution from U.S. patent 5,933,510, despite of its many advantages, fits only poorly for the acoustical design of a mobile terminal. In '510, a microphone housing includes apertures and a switch; the user may select with the switch, whether or not he or she is willing to use the microphone in omni- or unidirectional mode. Firstly, it is difficult to make the switch and sealing elements to comply with the decorative design desired for the terminal. Secondly, the design of a rotating sleeve for the microphone housing in the manner described in '510 is also extremely challenging.

[0011] It is an object of the invention to bring about a mobile terminal equipped with a sound receiving system having selectable directional characteristics.

Summary of the invention

[0012] This and other objectives of the invention are accomplished in accordance with the principles of the present invention by providing in accordance with any one of the independent patent claims.

[0013] A dual directional mode mobile terminal includes a microphone having been assembled in a housing. The housing includes a first and a second aperture, of which the first aperture is on the user side when the terminal is used as a handset, the first and second apertures forming acoustical conduits to the both sides of the microphone. The housing further includes sealing means for sealing the second aperture together with an external surface, when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a

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second acoustical directivity mode.

[0014] A method for manufacturing a dual directional mode mobile terminal, includes the steps of i) assembling a microphone into a housing, wherein the housing includes at least a first and a second apertures, of which the first aperture is designed to be on the user side when the terminal is used as a handset, the first and second apertures thus forming acoustical conduits to both sides of the microphone, and ii) assembling the housing with sealing means for sealing at least one of the apertures together with an external surface, when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a second acoustical directivity mode.

[0015] A mobile terminal which is equipped with an acoustoelectrical transducer arranged in a housing of the mobile terminal, wherein the housing comprises i) a user sided first aperture in communication with the acoustoelectrical transducer, ii) a second aperture also in communication with the acoustoelectrical transducer, and iii) sealing means for sealing the second aperture when the mobile terminal is placed on an external surface or object.

Brief Description of the Drawings

[0016] In the following, the invention and its preferred embodiments are described more closely referring to the examples shown in Figure 2-11 in the appended drawings, wherein:

Figure 1	shows a conventional microphone arrangement in a mobile terminal,
Figure 2	shows a dual-mode mobile terminal according to one aspect of the invention,
Figure 3	illustrates one embodiment of the present invention where acoustic sealing is enhanced by a second support member,
Figure 4	depicts the traverse design aspects in accordance with the invention,
Figure 5	further illustrates one embodiment of the invention,
Figure 6	illustrates a cut of a sealing member 203 from Figure 2,
Figure 7	illustrates the sealing member 203 seen from above,
Figure 8	shows the second acoustical mode closer,
Figure 9A and 9B	show how the directivity mode of the mobile terminal is changed according to one aspect of the present invention,
Figure 10	shows a first measurement result of the mobile terminal in the first directional mode, and
Figure 11	shows a second measurement result of the mobile terminal in the second directional mode.

[0017] Same reference numbers denote similar elements throughout the figures.

Detailed description of the invention

[0018] Figure 2 illustrates one embodiment of the present invention. The mobile terminal has a housing 200, and, as shown in Figure 1 described above, there is also an aperture 101 provided for the speech. The aperture 101 leads sound waves to a microphone 110 that is assembled inside the housing 200. In the housing 200 there is also a second aperture 202. The second aperture 202 is preferably located on the opposite side of the housing 200 so that the apertures 101 and 202 provide acoustic conduits leading to both sides of the microphone.

[0019] For sake of clarity, there are three points marked in the housing. PL and PR are the points where sealing means 203 are attached to the housing 200, and the point PA illustrates the point of the housing that supports the lower end of the housing against a flat surface 204 when the mobile terminal is tilted on said sealing means 203. The sealing means 203 are not drawn to scale; according to one aspect of the present invention their height is some millimetres. Depending on the aesthetic goals of the designer, sealing means 203 that are higher or lower can be used.

[0020] According to one embodiment of the present invention the mobile terminal can be placed on a table 205, for example, or other flat surface 204, so that the sealing means 203 together with the flat surface 204 seal the second aperture 202. The sealing of the second aperture 202 has the effect that the mobile terminal is in omni-directional mode where the microphone 110 receives voice signals mostly through the aperture 101 thus losing some of its directional sensitivity.

[0021] Normally, the microphone 110 is directionally sensitive when both apertures 101 and 202 are open. When both apertures 101, 102 are open, the sound waves coming from the background noise can be understood to cancel each other, i.e. the microphone shows unidirectional characteristics and is well suitable for to be used as a handset of a mobile terminal.

The uni- and omni-directional characteristics of the microphone 110 are herewith referred with the term acous-

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tical modes of the microphone.

[0022] Figure 3 shows another example of implementing the present invention. Instead of only having sealing means 303 such as a sealing member, the mobile terminal now also has a support member 304. In addition, the marked point PB illustrates the effective support point of the support member 304 that is comparable to point PA with reference to Figure 2.

[0023] One function of the support member 304 is to increase the pressure between the sealing means 303 and the flat surface 204. The sealing means 303 are located close to the second aperture 202 which is further located at the spot of the microphone. As the mobile terminal of Figure 2, the mobile terminal can as well be placed on a flat surface 204 thus giving omni-directional characteristics for the mobile terminal. The position PB which is also the position of the second support member 304 may be located based on the weight distribution of the mobile terminal. To the weight distribution influences the weight of the contents of the mobile terminal, i.e. the battery, different electrical parts, the display, and so forth. The PB can be positioned so that the torque caused by the weight of the mobile terminal causes a stronger or greater force on the sealing means 303 and so enhances the sealing made by the sealing means 303 and the flat surface 204. If the sealing means 303 are made of a soft material showing high elastic properties, a good sealing can be provided.

[0024] Figure 4 depicts a cross sectional cut of a mobile terminal. The terminal has a microphone 110 which is positioned inside the housing 200. In the housing 200 there are apertures 101 and 202. The apertures 101, 202 act as inlet holes which form acoustical conduits to both sides of the microphone 110. It is to be noted that the apertures 101, 202 need not to be on opposite sides of the mobile terminal; the only limiting condition is that the acoustical conduits lead to both sides of the microphone. Further, there are support and sealing means 203 at the bottom side of the housing. The design of the sealing means 203 may involve taking into account the curvature of the housing 200. In some mobile terminal models, the housing 200 is strongly curved and, therefore, it is impossible to provide a good enough sealing with sealing means 203 that do not take into account the shape of the bottom of the housing 200.

[0025] Further, Figure 5 illustrates the mobile terminal viewed from below. In the sealing means 203 that is upper part of the mobile terminal housing 200 the aperture 202 and the sealing means 203 are shown.

[0026] Figure 6 illustrates closer the sealing means 203. First, the sealing means 203 have a hole 600 for the acoustical inlet. The contact part 601 of the sealing means 203 has been only marked with a box on the right-hand side, but it preferably extends through the sealing means 203. For example, if the sealing means 203 have the form of a ring like in Figure 5, the part 601 extends around the bottom of the ring.

[0027] Further, aperture part 603 has similarly been marked with a box. The acoustical signals coming from outside the mobile phone in order to reach the inlet hole 600 and to further proceed into the microphone 110 traverse beyond this part. In its design one must take into account the acoustical considerations: Special curvature may be needed in order to provide a proper distortion free acoustical transmission characteristics.

[0028] The sealing means 203 are connected to the mobile terminal using the aperture part 605 of the sealing means 203. On the left-hand side the height of the sealing means 203 is denoted with h' and on the right-hand side with h. The difference (h'- h) between h' and h is denoted with Δ. If the microphone is located as usually at the bottom part of the mobile terminal and if the tilting of the housing because of the height difference of the sealing means and the bottom part of the housing is like in Figure 2, h is larger than h'. The tilting may therefore depend on the distance between PA and PL, and partially as well on the distance between PL and PR. In the case of a mobile terminal with a support element like in Figure 3 there is no need for this kind of asymmetry and h' can equal h. The size of Δ may be adapted to make the sealing proper taking into account the tilting angle of the between the bottom part of the housing and the surface where the mobile terminal is lying.

[0029] Another possibility to achieve an asymmetric elastic behaviour is to use elastic materials asymmetrically, i.e. a more elastic material on the h' side and a stiffer material on the opposite side. This can in principle be achieved with e.g. block copolymers where the ratios of different polymer materials exhibit small changes along the bottom part of the sealing means. In this way it is also possible to ensure a good enough sealing. Another option, for example, is to use some other material that has spatially non uniform elasticity constant.

[0030] Figure 7 shows the sealing means from bottom. 600 is the inlet and 601 is the contact part. The profile of the aperture part 603 with its curvature is not visible in the figure.

[0031] Figure 8 presents the mobile terminal when the second acoustical directivity mode is activated. The mobile terminal 102 is placed on a flat surface 204. In the figure only the portion of the mobile terminal which illustrates the housing means and their vicinity is shown. The sealing means 203 have a contact with the flat surface 204 at aperture points PLB and PRB. Preferably, the sealing means extend from the housing of the mobile phone in such a way that in the figure left-hand side contact point PL acts as the outermost aperture point for the sealing means at part PLB and, respectively, the point PR at as the contact point of the sealing means for the point PRB.

[0032] The housing 200 is usually constructed from two parts; 102F is the front part and 102B is the back part. Between the front and back part 102F, 102B of the housing 200 there is a microphone 110. In the front part 102F at the spot of the microphone 110 there is at least one opening for the aperture 101 and on the back side

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part of the housing there is correspondingly at least one aperture 202 leading to the bottom side of the microphone.

[0033] Between the housing and the microphone there is sealing on both sides of the microphone 110. Preferably, the sealing is obtained with the help of an elastic gasket, such as a sealing ring. The first purpose of the sealing is to enable the propagation of sound from the exterior of the housing to the microphone, disabling propagation of the sound inside the housing, whereas the second purpose is to reduce the distortions caused by the loudspeaker of the mobile terminal.

[0034] Below the microphone 110 there is a resonance cavity ACB which is consists of smaller cavities formed by i) space limited by the microphone 110, possible sealing between the microphone, and bottom part 102 of the housing 200, and ii) acoustically well conductive part of the housing 200 including at least one aperture 202, sealing means 203, and the flat surface S. Respectively, another resonance cavity ACF is formed in between the microphone and the front part of the housing.

[0035] As already discussed with reference to Figures 2 and 6, the mobile phone may be tilted. To serve this purpose the sealing means are designed in the manner described with reference to Figure 6, the distance between PL and PLB may be shorter than the distance between PR and PRB. This means that the PL, PLB part could be more elastic than the PR and PRB so that the PL, PLB part of the sealing means compresses more. The aperture between the flat surface 204 and the sealing means 203 shall be preferably good in order to enhance the omni-directional mode. It is possible that if the sealing is not proper the direction and sensitivity of microphone is reduced to correspond the unidirectional microphone.

[0036] The tilting angle α corresponds to the tilting of the mobile terminal with respect to the flat surface 204. In practice, it may get almost any target value, the target value being thus between a very small angle and a substantially large angle. The magnitude of α has an impact on the aesthetic design of the mobile terminal, and, secondly, it may influence on the contact force between the contact part 601 and the flat surface 204 thus leading to further elasticity considerations. The more the mobile terminal is tilted, the smaller the contact force in the support means is, so requiring more elastic material.

[0037] Figure 9A illustrates another way to realise the invention. The mobile terminal has a housing 200 which includes a battery cover 901. The battery cover may be removable; usually the mobile terminal battery is stored under the cover. For some manufacturers the cover and the battery are combined to a single element. The aperture 202 is marked with a dot in the figure. When the inlet hole is open the mobile phone is in unidirectional directivity mode.

[0038] Figure 9B shows the aperture 202 covered with removable sealing means 1001 according to one aspect of the present invention.

[0039] The removable sealing means 1001 can be stored under the battery cover or back cover 901. Preferably, the aperture 202 and the sealing means 1001 are designed in such a way that the sealing means are tightly matched with the aperture 202. This may be realised, for example, by supplying the sealing means with attaching means and sealing means, so that the sealing means 1001 correspond partially to a flat surface 204. As shown in Figure 8, this mode of the invention is particularly usable when the table 205 or other surface is not essentially flat so that it would not provide a uniform enough flat surface 204 for the sealing means 203 to match at points PLB and PRB shown in Figure 8. Further, the external surface can be used for causing this kind of extra pressure for pushing the sealing means 203 stronger against the housing 200 of the mobile terminal, because the sealing means 1001 would not necessarily be able to cause the sealing to be tight enough hence the weight of the mobile terminal can be used for this purpose to increase the sealing.

[0040] Figure 10 illustrates a measurement result of the spectral energy density versus the frequency of a signal output at a reference point close to the telephone. The measurement has been performed for the inventive mobile phone assembly in unidirectional mode, and in Loudness Rating Guarding Position LRGP in accordance with the standard 3G TS 26.132 (TSG SA; Codec Working Group; Terminal Acoustic Characteristics for Telephony).

[0041] The energy spectrum measured is presented as observed by the codec of the mobile terminal. The upper piecewise linear curve corresponds to the maximal allowed spectral density values, whereas the lower piecewise linear curve illustrates the minimum acceptable spectral energies. As can be seen from the measurement results, the observed spectra lies clearly between the allowed values.

[0042] Figure 11 shows the acoustic behaviour of the same mobile terminal when it is in the second acoustical directivity mode. The spectral energy density shows again that the values are allowed, i.e. that the spectra lies clearly between the agreed limits.

[0043] Both measurements were conducted with a the mobile terminal lying head up on the table, positioned 50 cm from the reference mouth and with 30° azimuth. For the measurements the mobile terminal input gain was not adjusted, though this is not visible in the figure. In practice, because of the different resonance cavities caused by the differences in the acoustical apertures when the mobile terminal is in different acoustical modes, the gain must be adjusted between the modes. The adjustment may be done automatically by detecting the measured signal strength in the codec.

[0044] To summarise, according to one aspect of the present invention, the mobile phone can be made to show dual mode characteristics. First, a microphone is assembled into a housing that includes at least two inlet

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holes. At least one of the apertures is designed to be on the user side when the terminal is used as a handset, and the apertures together form acoustical conduits to the both sides of the microphone. The housing is further assembled to include sealing means; with the sealing means at least one of the inlet holes may be sealed together with an external surface, such as a table or any other flat surface. When the mobile terminal is placed on said flat surface, one acoustical aperture is thus closed and the mobile terminal shows acoustical omni-directional characteristics as a second acoustical directivity mode.

[0045] According to one further aspect of the present invention, the assembling step may include the assembling of an elastic gasket. Preferably, said elastic gasket is asymmetric, and the height of the lower part of the gasket being smaller than the height of the higher part of the gasket, the difference being adapted to make the sealing proper taking into account the elevation angle between the bottom of housing and the surface.

[0046] According to still another aspect of the present invention, in addition to the sealing means, the housing may further include a support member to increase the pressure between the sealing means and the surface.

[0047] Although the invention was described above with reference to the examples shown in the appended drawings, it is obvious that the invention is not limited to these, but it may be modified by those skilled in the art without departing from the scope and the spirit of the invention. For example, instead of a microphone, any other transducer or sound detector may be used in connection with the embodiments of the inventive sealing mechanism.

Reference numbers used in the description:

[0048]

100	mobile terminal
101	aperture
102	housing
110	microphone
200	housing
202	second aperture
203	sealing means
204	flat surface
205	table
303	sealing means
304	support member
600	hole for the acoustic inlet
601	contact part
603	aperture part
901	battery cover
1001	removable sealing means

Claims

1. A dual directional mode mobile terminal including a microphone (110), said microphone (110) having been assembled in a housing (200),
characterised in that
   - the housing (200) includes at least a first and a second aperture (101, 202), of which the first aperture (101) is on the user side when the terminal is used as a handset, the first and second apertures (101, 202) forming acoustical conduits (ACF, ACB) to the both sides of the microphone (110); and
   - the housing (200) further includes sealing means (203, 303, 1001) for sealing the second aperture (202) together with an external surface (204), when the mobile terminal is placed on said surface, thereby enabling in the mobile terminal a second acoustical directivity mode.

2. A mobile terminal according to claim 1,
characterised in that
the sealing means (203) include an elastic gasket.

3. A mobile terminal according to claim 2,
characterised in that
the elastic gasket is asymmetric, the height h' of a lower part of the gasket being smaller than the height h of a higher part of the gasket, the difference $\Delta = h' - h$ being adapted to make the sealing proper taking into account an elevation angle (α) between a backside surface (102B) of the housing (200) and the external surface (204).

4. A mobile terminal according to claim 1,
characterised in that
in addition to the sealing means (303), the housing (200) further includes a support member (304) to increase the pressure exerted on the sealing means (303) by the external surface (204).

5. A mobile terminal according to claim 1,
characterised in that
the sealing means (1001) further have means adapted for connecting the sealing means to the inlet hole (202) in an acoustically proper way.

6. A mobile terminal according to claim 5,
characterised in that
the mobile terminal has means (200) for storing the sealing means (1001).

7. A method for manufacturing a dual directional mode mobile terminal, the method
characterised in that
it includes the steps of:

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- assembling a microphone (110) into a housing (200), wherein the housing (200) includes at least first and second apertures (101, 202), of which a first aperture (101) is designed to be on the user side when the terminal is used as a handset, the first and second apertures thus forming acoustical conduits to both sides of the microphone (110); and
- assembling the housing (200) with sealing means (203) for sealing the second aperture (202) holes together with an external surface (204), when the mobile terminal is placed on said external surface (204), thereby enabling in the mobile terminal a second acoustical directivity mode.

8. A method according to claim 7, **characterised in that** the assembling step includes the assembling of an elastic gasket.

9. A method according to claim 8, **characterised in that** said elastic gasket is asymmetric, the height h' of the lower part of the gasket being smaller than the height h of the higher part of the gasket, the difference Δ = h'- h being adapted to make the sealing proper taking into account the elevation angle (α) between the bottom (102B) of housing and the external surface (204).

10. A method according to claim 7, **characterised in that** in addition to the sealing means (303), the housing further includes a support member (304) to increase the pressure between the sealing means (303) and the surface (204).

11. A method according to claim 7, wherein the sealing means (1001) are further adapted to be connected to the inlet hole (202) in an acoustically proper way.

12. A method according to claim 11, wherein the sealing means (1001) are stored in the mobile phone, and when used, manually placed into the aperture (202).

13. A mobile terminal equipped with an acoustoelectrical transducer (110) arranged in a housing (200) of the mobile terminal, **characterised in that** the housing (200) comprises:

- a user sided first aperture (101) in communication with the acoustoelectrical transducer (110);
- a second aperture (202) also in communication with the acoustoelectrical transducer (110); and
- sealing means (203) for sealing the second aperture (202) when the mobile terminal is placed on an external surface (204) or an object.

14. A mobile terminal according to claim 13, wherein the sealing means (203) are responsive to an external force or pressure.

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FIG 1 Prior Art

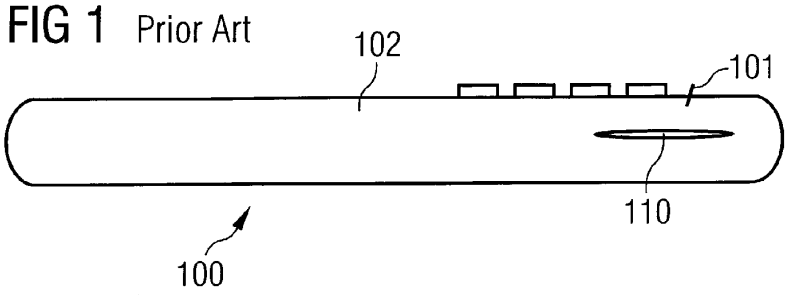


FIG 2

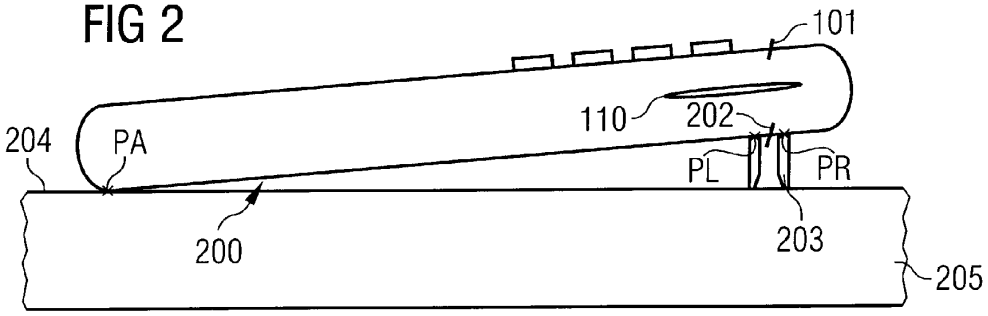


FIG 3

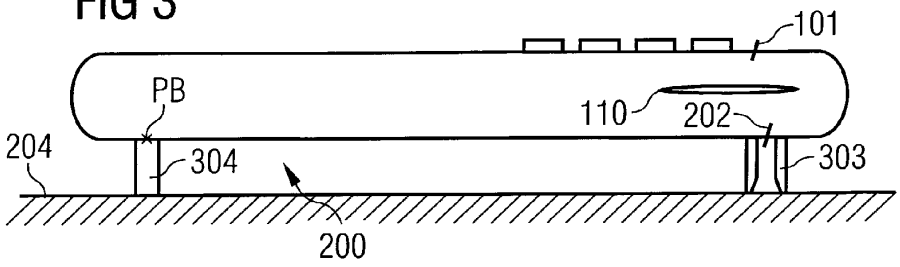
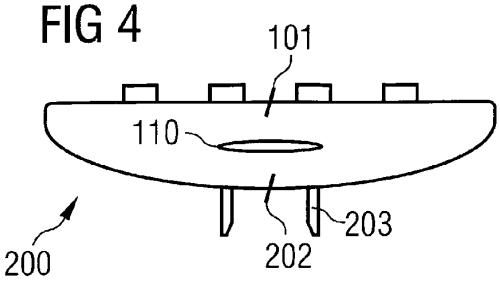


FIG 4



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FIG 5

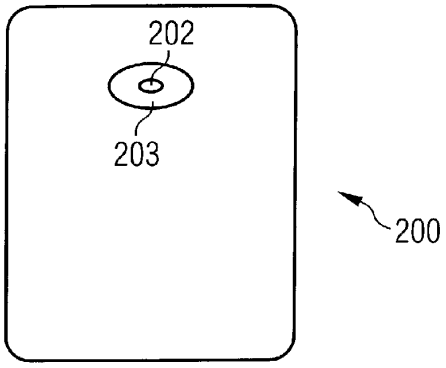


FIG 6

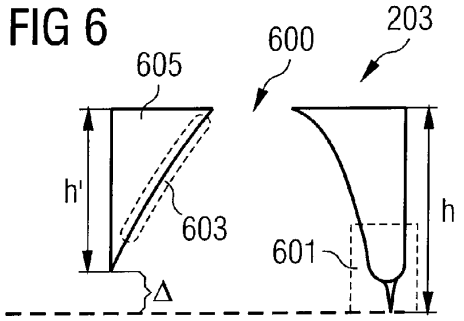


FIG 7

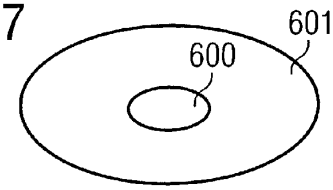
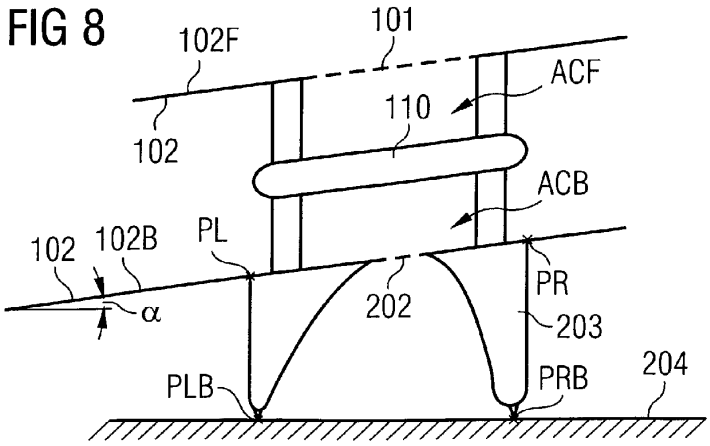


FIG 8



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FIG 9A

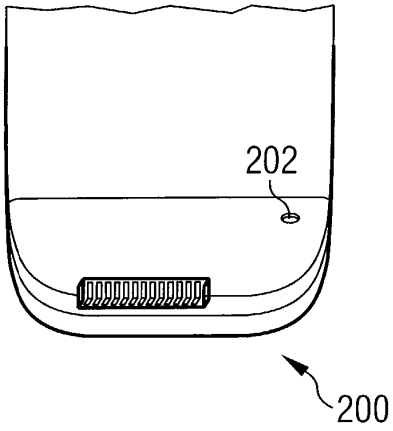
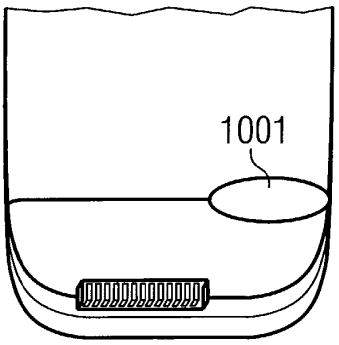


FIG 9B



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FIG 10

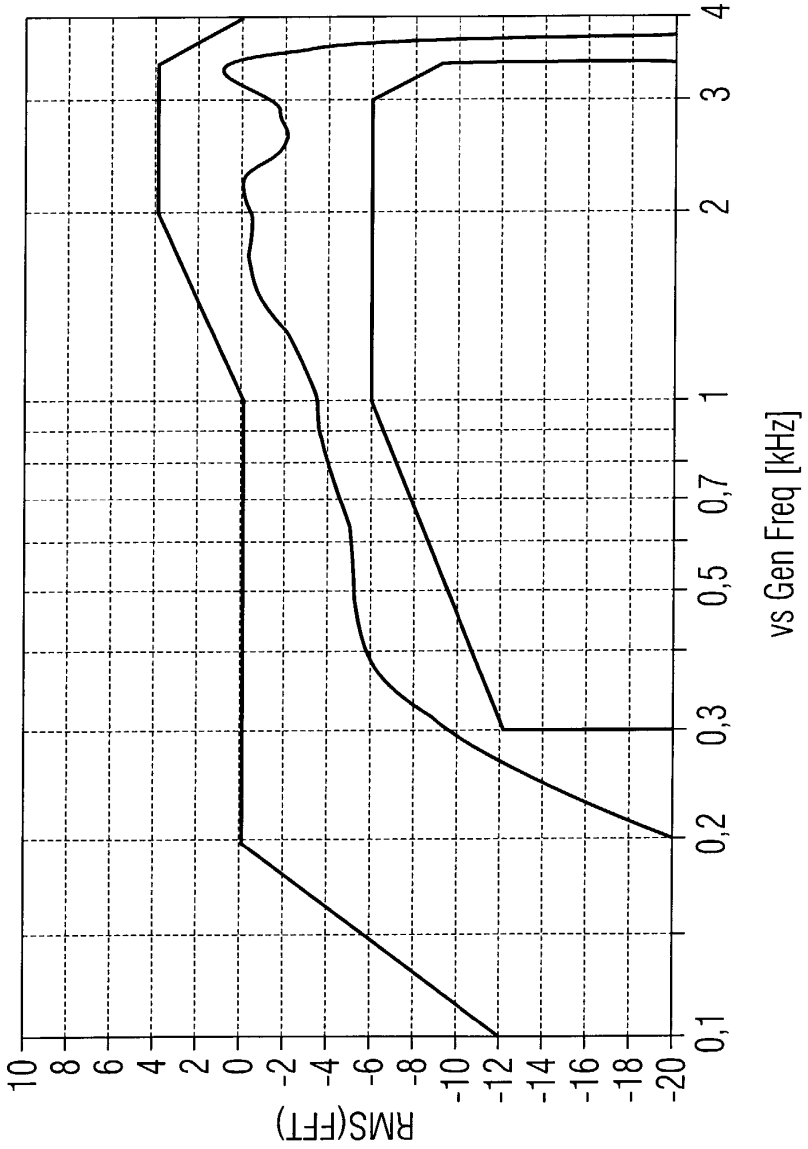
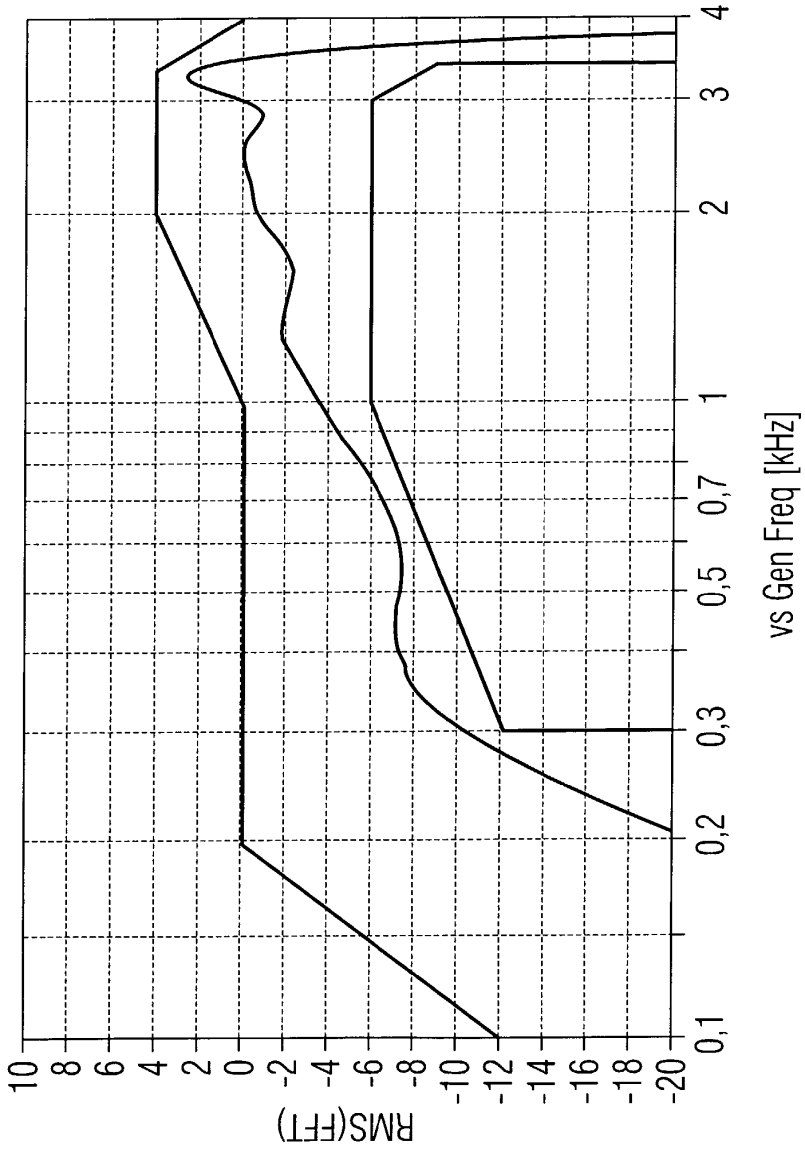

RMS(FFT)
10
8
6
4
2
0
-2
-4
-6
-8
-10
-12
-14
-16
-18
-20
0,1
0,2
0,3
0,5
0,7
1
2
3
4
vs Gen Freq [kHz]

FIG 11


RMS(FFT)
10
8
6
4
2
0
-2
-4
-6
-8
-10
-12
-14
-16
-18
-20
0,1
0,2
0,3
0,5
0,7
1
2
3
4
vs Gen Freq [kHz]

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 02 0319

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
A	US 4 142 072 A (BERLAND OLE) 27 February 1979 (1979-02-27) * the whole document * ---	1-14	H04R25/00
A	DE 100 50 559 C (FAITA EBERHARD) 4 April 2002 (2002-04-04) * the whole document * ---	1-14	
A	EP 0 984 666 A (RESISTANCE TECHNOLOGY INC) 8 March 2000 (2000-03-08) * abstract * * claims 1,2 * ---	1-14	
A	WO 01 69964 A (RESISTANCE TECHNOLOGY INC ;DARBUT ALEXANDER L (US)) 20 September 2001 (2001-09-20) * page 2, line 26 - page 3, line 9 * * abstract * -----	1-14	

TECHNICAL FIELDS SEARCHED (Int.Cl.7)

H04R

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	31 March 2003	Dionisi, M

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
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P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

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ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 02 02 0319

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report. The members are as contained in the European Patent Office EDP file on
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31-03-2003

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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(11) EP 1 519 625 B1

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H04R 25/00 *(2006.01)*

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(54) External ear canal voice detection

Sprachdetektion im Gehörgang

Détection de la voix dans le conduit auditif externe

(84) Designated Contracting States:
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WO-A-03/073790
WO-A-2005/004534
WO-A-2005/125269

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Printed by Jouve, 75001 PARIS (FR)

EP 1 519 625 B1

Description

[0001] This document generally relates to ear-level communication systems and particularly, but not by way of limitation, to such systems communicating with remote devices via full duplex audio.

[0002] Ear-level communication systems are used for hearing impaired individuals to use a communication device such as a cellular phone or other devices transmitting audio signals. They also allow private, hand-free use of such a communication device by individuals with normal hearing.

[0003] An ear-level communication system generally includes tethered or wireless headsets with a microphone boom or a microphone within a cord of the headset to detect voice of the person wearing the headsets. An alternative way to detect voice is via bone connection. Examples include a "clam" shell device that couples to the helix and a custom carmold with a vibration detector on the outer shell. Voice is picked up by detecting bone vibration caused by, and representative of, the voice of the person wearing the device.

[0004] Bi-directional communication has been achieved by a simplex system including a device in the ear canal that both transmits sound to and picks up sound within the ear canal. The device is tethered to a push-to-talk (PIT) system such as a walkie talkie. It is not full duplex and therefore requires interaction by the user in order to switch from listen mode to talk mode.

[0005] Voice operated exchange (VOX) is used to conserve energy when voice is not being detected or transmitted. For example, a headset system includes two ear pieces - for full duplex communication with one earpiece for transmitting and the other for receiving voice - and uses ear canal voice detection to trigger a VOX circuit in a control unit wired to the headset. The system requires a headset with two headphones and wires connecting the headphone to a control unit such as a device clipped on belt. In another example, a full duplex system includes a single headphone to transmit and receiving voice, again with a VOX circuit in a control unit wired to the headphone.

[0006] Such ear-level communication systems are likely to be worn by a person for long periods of time. Users with hearing loss may use such a system on a regular basis, both as a conventional hearing aid allowing communication directly with a person and as a hearing aid allowing communicating through another device. Users with normal hearing may wear such a system to be communicatively connected to another person or facility without the need to hold a device at any time. For appearance or secrecy reasons, minimal visibility is generally desirable. Wired connection between a headset or earpiece and a controller makes the system easily visible and is cumbersome to users who need to communicate while being physically active.

[0007] Thus, there is a need for an ear-level communication system that is not easily visible. A need related to miniaturization is the longevity of the system between battery replacements or recharges.

[0008] WO 98/45937A discloses a communications apparatus comprising ear units, a control unit and a radio transceiver.

[0009] Ear level full duplex audio communication systems each include one or two ear attachments devices, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, that wirelessly communicates to a remote device. Such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. Such a system also functions as an ear piece wirelessly extended from the remote device that allows an individual to privately communicate with or through the remote device without the need of holding the device.

[0010] WO 03/073790A discloses a system adapted for use in at least one of a first ear having a first ear canal and a second ear having a second ear canal comprising: a first ear-level device configured to be one of an in-the-ear device and a behind-the-ear device, the first ear-level device including: a first microphone adapted for use in one of the first ear canal and the second ear canal, and adapted to receive an occluded sound within the one of the first ear canal and the second ear canal; and a wireless receiver, connected to a first processor, to receive a first wireless signal representing a remote sound; wherein the first processor, is connected to the first microphone, and is adapted to process the occluded sound and the remote sound; the first ear level device further including a speaker adapted for use in the one of the first ear canal and the second ear canal and adapted to transmit the processed remote sound to the one of the first ear canal and the second ear canal; a wireless transmitter, connected to the first processor; a first earmold housing at least the first microphone and the speaker, at least a portion of the first earmold configured for placement in the one of the first ear canal and the second ear canal; and a voice operated exchange (VOX) circuit connected to the first processor.

[0011] In the present invention, the wireless transmitter is adapted to receive the processed occluded sound from the first processor and transmit a second wireless signal representing the occluded sound; and the VOX circuit includes an occluded sound gating module adapted to gate the occluded sound on, and to gate the remote sound off, when the occluded sound is detected while the remote sound is not detected, and a remote sound gating module adapted to gate the remote sound on, and to gate the occluded sound off, when the incoming remote sound is detected while the occluded sound is not detected.

[0012] In a preferred embodiment, a system includes an ITE module and a BTE module attached to the ITE module. The ITE module includes a microphone for use in an ear canal. The microphone receives an occluded sound from within the ear canal. The BTE module includes a processor and a wireless transmitter. The processor processes the occluded sound. The wireless trans-

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mitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0013] In a preferred embodiment, a system includes a first ear-level device and a second ear-level device for use in two opposite ears. The first ear-level device includes a microphone, a first processor, and a wireless transmitter. The microphone is for use in one ear canal to receive an occluded sound from within that ear canal. The first processor converts the occluded sound to an electrical signal. The wireless transmitter receives the electrical signal and transmits an outgoing wireless signal representing the occluded sound. The second ear-level device includes a wireless receiver, a second processor, and a speaker. The wireless receiver receives an incoming wireless signal representing a remote sound. The second processor converts the incoming wireless signal to the remote sound. The speaker is for use in the opposite ear canal to transmit the remote sound to that ear canal.

[0014] The present invention also relates to a method comprising: detecting an occluded sound using a first microphone adapted for use in an ear canal; detecting an incoming radio signal representing a remote sound from a remote device; starting a voice operated exchange (VOX) housed within an ear-level communication device when at least one of the occluded sound and the incoming radio signal is detected if the VOX is not already on, the ear-level communication device being one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device; gating the occluded sound on, and gating the remote sound off, when the occluded sound is detected while the incoming radio signal is not detected; and gating the remote sound on, and gating the occluded sound off, when the incoming radio signal is detected while the occluded sound is not detected.

[0015] Preferred embodiments of the present invention will now be described by way of example only and with reference to the accompanying drawings, in which:

FIG. 1 is an illustration of an embodiment of an ear-level communication system and potions of an environment in which it is used. This ear-level communication system uses a single ear-level device.

FIG. 2 is an illustration of an embodiment of another ear-level communication system and portions of an environment in which it is used. This ear-level communication system uses a pair of ear-level devices.

FIG. 3A is an illustration of one exemplary exterior configuration of an in-the-ear (ITE) device used as the single ear-level device of FIG 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 3B is an illustration of one exemplary exterior configuration of an in-the-canal (ITC) device used as the single ear-level device in FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 3C is an illustration of one exemplary exterior configuration of a completely-in-the-canal (CIC) device used as the single-ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device of FIG. 1.

FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single car-level device of FIG. 1.

FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the pair of ear-level devices of FIG. 1B.

FIG. 6 is an illustration of one exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 7 is a block diagram showing one embodiment of the circuit of the ear-level device shown in FIG. 6 used as the single ear-level device of FIG. 1.

FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as the pair of ear-level devices of FIG. 2.

FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device.

FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication between the single ear-level device and a remote device.

[0016] The following embodiments are described in sufficient detail to enable those skilled in the art to practice the invention, and it is to be understood that the embodiments may be combined, or that other embodiments may be utilized and that structural, logical and electrical changes may be made without departing from the scope of the present invention, as defined by the appended claims.

[0017] It should be noted that references to "an", "one", or "various" embodiments in this disclosure are not necessarily to the same embodiment, and such references contemplate more than one embodiment.

[0018] This document discusses, among other things, ear-level full duplex audio communication systems for communicating with remote devices. These systems each include one or two ear-level devices each worn in or about an ear, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, wirelessly coupled to a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. When used as a hearing aid, such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. When being used as an ear piece wirelessly extended from the remote device,

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such system allows an individual with normal hearing to privately communicate with or through the remote device without the need of holding the device or wearing any device wired to the remote device.

[0019] In this document, "voice" includes to the sound of speech made by a person, and sound generally includes an audible signal that includes voice and other signals that can be heard and/or detected by an audio detector. Unless particularly indicated, for example, when voice is distinguished from sound by speech recognition, "voice" and "sound" are used interchangeably throughout thus document because they are not practically separated. For example, detection of a sound refers to the same action as detection of a voice because, while the purpose is to detect the voice, other sounds are picked up as well.

[0020] FIG. 1 is an illustration of an embodiment of an ear-level communication system 100 and portions of an environment in which it is used. System 100 includes an ear-level device 110 attached to an ear 101A. Bar-level device 110 communicates with a remote device 190 through a wireless telemetry link 195. The other ear 101B is free of device attachment. Ear-level device 110 is a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of ear-level device 110 is voice activated. That is, ear-level device 110 is activated only when it detects a need to transmit or receive a sound.

[0021] Ear-level device 110 is a full duplex audio device that allows two-way simultaneous conversation between ear 101A and remote device 190. In one embodiment, when being worn by a person having ear 101A, ear-level device 110 picks up sound from the ear canal of ear 101A and delivers sound to the same ear canal. It detects the occluded sound from the ear canal when the person speaks and transmits to the same ear canal a sound received from remote device 190 and/or a sound picked up from the environment surrounding the person. In one embodiment, as illustrated in FIG. 1, ear-level device 110 is configured for right ear attachment. In an alternative embodiment, ear-level device 110 is configured for left ear attachment.

[0022] FIG. 2 is an illustration of an embodiment of an ear-level communication system 200 and portions of an environment in which it is used. System 200 includes a pair of ear-level devices 210A, attached to an ear 101 A, and 210B, attached to ear 101B. Ear-level devices 210A and 210B communicate with a remote device 290 through a wireless telemetry link 295. Ear-level devices 210A and 210B are each a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of each of ear-level devices 210A and 210B is voice activated. That is, ear-level devices 210A and 210B are each activated only when a need to transmit or receive a sound is detected by the device itself.

[0023] In one embodiment, ear-level device 210A is an audio transmitter that picks up sound from the ear canal of ear 101A, and ear-level device 210B is an audio receiver that delivers sound to the same ear canal. Ear-level device 210A is activated when it detects a sound from the ear canal. Ear-level device 210B is activated when it detects a signal from remote device 290. When both are being worn by a person, ear-level devices 210A and 210B supports full duplex audio allowing two-way simultaneous conversation between the person and remote device 290. In one embodiment, as illustrated in FIG. 2, ear-level device 210A is configured for right ear attachment, and ear-level device 210B is configured for left ear attachment. In an alternative embodiment, ear-level device 210A is configured for left ear attachment, and ear-level device 210B is configured for right ear attachment.

[0024] FIGS. 3A, 3B, and 3C are illustrations of exemplary exterior configuration of ear-level device 110 or one of ear-level devices 210A and 210B. For the purpose of this document, only a device attached to a right ear is shown for each exemplary exterior configuration. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a device configured for right ear attachment or a device configured for left ear attachment. Generally, a device configured for right car attachment has an exterior configuration that is substantially symmetrical to that of a device of the same type configured for left ear attachment. In system 100, a single ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of car-level device 210A and 210B is a device configured for right ear attachment, while the other is a device configured for left ear attachment.

[0025] An occluded sound 305, an ambient sound 306, and a remote sound 307 are illustrated in each of FIGS. 3A, 3B, and 3C. Remote sound 307 is represented by a wireless radio signal transmitted to ear-level device 110. In this document, "occluded sound" includes occluded sound 305, "ambient sound" includes ambient sound 306, and "remote sound" includes remote sound 307.

[0026] FIG. 3A is an illustration of an exemplary exterior configuration of an in-the-ear (ITE) device 310A used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3A, ITE device 310A is attached to an ear 101 with a pinna 302 and an ear canal 303. ITE device 310A includes a portion inserted into ear canal 303 and another portion fit into the cavity formed by pinna 302.

[0027] FIG. 3B is an illustration of an exemplary exterior configuration of an in-the-canal (ITC) device 310B used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3B, ITC device 310B is attached to ear 101 with pinna 302 and ear canal 303. ITC device 310B is a form of ITE device that also includes a portion inserted in ear canal 303 and another portion fit into the cavity formed by pinna 302, but the latter portion is smaller than that of ITE device 310A. Its overall

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size is therefore generally smaller than that of ITE deviee 310A.

[0028] FIG. 3C is an illustration of an exemplary exterior configuration of a completely-in-the-canal (CIC) device 310C used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3C, CIC device 310C is attached to ear 101 with pinna 302 and ear canal 303. CIC device 310C is a form of ITC device but includes only a portion that is completed inserted into ear canal 303. Its overall size is therefore generally smaller than that of ITC device 310B.

[0029] In one embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a custom earmold. In another embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0030] The choice of a device type, such as from the types shown in FIGS. 3A, 3B, and 3C, depends on the need of individual users. For example, ITE device 310A has a relatively large size to accommodate a relatively large amplifier to serve the need of individuals suffering severe hearing loss. The relatively large size can also be utilized to house a relatively large size battery. On the other hand, CIC device 310C is almost invisible, while the room for the circuit and the battery is very limited. Generally, the selection of the device type is a compromise among considerations such as the user's degree of hearing loss (if applicable), ear canal size, desire of invisibility, desired longevity between battery replacements or recharges, and cost.

[0031] FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device 110. The circuit is housed in an earmold configured for use as one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0032] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an car canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not "isolated" when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, end wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110.

[0033] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected

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by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440.

[0034] For the purpose of description, each of the gating modules is "on" when sound is "gated on" and "off' when the sound is "gated off" A sound is "gated on" when it is allowed to pass and be processed, and is "gated off' when it is blanked or substantially attenuated. Blanking refers to a substantially complete blockage of a sound, or in other words, that a detected sound is practically ignored by processor 460 such that it does not cause any echo or ringing that is audible by an ear. Substantial attenuation refers to an attenuation after which the attenuated sound does not cause any echo or tinging having an intolerable intensity.

[0035] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay.

[0036] Occluded sound gating module 441 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected occluded sound with a predetermined threshold for the occluded sound. When the amplitude of the detected occluded sound exceeds the threshold for the occluded sound, and when controller 431 permits, occluded sound gating module 441 is turned on. When the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off. In one embodiment, after the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off after a predetermined delay.

[0037] In one embodiment, remote sound gating module 451 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a remote sound with a predetermined threshold for the remote sound. When the amplitude of the detected remote sound exceeds the threshold for the remote sound, and when controller 431 permits, remote sound gating module 451 is turned on. When the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off. In one embodiment, after the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off after a predetermined delay.

[0038] In another embodiment, remote sound gating module 451 includes a switch or attenuator driven by the detection of the incoming wireless signal representing the remote sound. Remote sound gating module 451 is turned on upon detection of the incoming wireless signal when controller 431 permits, and turned off when the incoming wireless signal is no longer detected. In one embodiment, remote sound gating module 451 is turned off after the incoming wireless signal is absent for a predetermined period.

[0039] Controller 431 coordinates the on/off states of all the gating modules so the sounds are processed in an orderly fashion. Even if a sound is detected, the gating module corresponding to that sound is turned on only when controller 431 permits. In one embodiment, assuming that sounds A and B are gated by gating modules A and B, respectively, if both sounds A and B are detected, whichever sound is detected earlier is gated on. That is, if sound A exceeds its threshold first, gating module A is turned on and remain on until sound A drops below threshold A. Then, gating module B is turned only if sound B exceeds its threshold after gating module A has been turn off.

[0040] In one embodiment, controller 431 coordinates the on/off states of all the gating modules according to predetermined or programmed gating coordination rules. According to one exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows only the transmission of the remote sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. According to another exemplary rule, ambient sound gating module 421 is on, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows the transmission of both the remote sound and the ambient sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. This also prevents the ambient sound from being transmitted to remote device 190. According to yet another exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is on, and remote sound gating module 451 is off. When the person wearing ear-level device 110 speaks, both external microphone 420 and ear canal microphone 440 detect the voice. The voice as detected by ear-level

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device 110 is first transmitted to remote device 190. Without the gating modules (or when ambient sound gating module 421 and occluded sound gating module 441 are both on), the same voice as detected by external microphone 420 is transmitted to ear canal 303, and therefore again detected by ear canal microphone 440, and again be transmitted to remote device 190. This rule allows only the transmission of the occluded sound to remote device 190, and prevents the same sound from being transmitted to remote device 190 twice. Other rules are applied as a person skilled in the art should see fit based on an understanding after reading this entire document.

[0041] FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as each of ear-level devices 210A and 210B. The circuit of each of ear-level devices 210A and 210B is housed in an earmold configured for use as of one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0042] Ear-level device 210A is an audio transmitter that includes an ear canal microphone 540, a processor 560A, a wireless transmitter 550A, an antenna 555A, a transmitter VOX circuit 530A, and a battery 580A. Ear canal microphone 540 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 210A. Processor 560A converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 290. In one embodiment, processor 560A includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560A includes a speech recognition module used to enhance the sound transmitted from ear-level device 210A. Wireless transmitter 550A and antenna 555A form a telemetry interface supporting signal transmission from ear-level device 210A to remote device 290. Wireless transmitter 550A receives the electrical signal representing the sound picked up from the ear canal from processor 560A and transmits an outgoing wireless signal representing the same sound to remote device 290. VOX circuit 530A activates a major portion of ear-level device 210A only when sound is detected by ear canal microphone 540. In one embodiment, VOX circuit 530A includes a voice-controlled switch connecting between battery 580A and the portions of ear-level device 210A that are voice activated. The switch is driven by the sound detected by ear canal microphone 540. It is turned on when sound detected by ear canal microphone 540 exceeds a predetermined threshold. In one embodiment, VOX circuit 530A activates processor 560A when the sound is detected. In one further embodiment, VOX circuit 530A also activates wireless transmitter 550A when the sound is detected. Battery 580A supplies the power needed for the operation of ear-level device 210A. In one embodiment, battery 580A is a rechargeable battery that can be recharged without being taken out from ear-level device 210A.

[0043] Ear-level device 210B is an audio receiver that includes an external microphone 520, an ear canal speaker 570, a processor 560B, a wireless receiver 550B, an antenna 555B, a receiver VOX circuit 530B, and a battery 580B. External microphone 520 picks up ambient sound such that the person wearing ear-level devices 210A and 210B is not "isolated" when communicating to remote device 290. It allows that person to talk to a nearby person directly, without the need to take off ear-level devices 210A and/or 210B. Bar canal speaker 570 transmits sound received from remote device 290 and/or external microphone 520 to the same ear canal. Processor 560B converts the signals received from remote device 290 and/or external microphone 520 to a sound audible to the person wearing ear-level device 210B. In one embodiment, processor 560B includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560B includes a speech recognition module used to enhance the audio signals received by ear-level device 210B. Wireless receiver 550B and antenna 555B form a telemetry interface supporting signal transmission from remote device 290 to ear-level device 210B. Wireless receiver 550B receives an incoming wireless signal representing a remote sound from remote device 290 and transmits an electrical signal representing the remote sound to processor 560B. VOX circuit 530B activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 520 and wireless receiver 550B. In one embodiment, VOX circuit 530B includes a voice-controlled switch connecting between battery 580B and the portions of ear-level device 210B that are voice activated. The switch is driven by the sound detected by at least one of external microphone 520 and wireless receiver 550B. It is turned on when sound detected by external microphone 520 exceeds a predetermined threshold and/or when wireless transceiver 550B receives the incoming wireless signal representing the remote sound. In one embodiment, VOX circuit 530B activates processor 560B when the sound is detected. In one further embodiment, VOX circuit 530B also activates ear-level speaker 570 when the sound is detected. Battery 580B supplies the power needed for the operation of ear-level device 210B. In one embodiment, battery 580B is a rechargeable battery that can be recharged without being taken out from ear-level device 210B.

[0044] System 200 accommodates a larger overall circuit size without necessarily increasing the size of each ITE device. While two separate ear-level devices are required, because system 200 includes separate audio transmitter (ear-level device 210A) and receiver (ear-level device 210B) disposed in the ear canals of opposite ears, ear canal speaker 570 and ear canal microphone 540 do not form a loop causing echo and ringing.

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[0045] FIG. 6 is an illustration of an exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device 610 used as ear-level device 110 or one of ear-level devices 210A and 210B. BTE device 610 is attached to ear 101 with pinna 302 and ear canal 303. In one embodiment, BTE device 610 is used when none of ITE device 310A, PTC device 310B, and CIC device 310C provides sufficient space accommodating the required circuit and battery sizes. For the purpose of this document, only a device attached to a right ear is shown for each exterior structure. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a BTE device configured for right ear attachment or a BTE device configured for left ear attachment. Generally, a BTE device configured for right ear attachment has a substantially symmetrical appearance to a device of the same type configured for left ear attachment In system 100, ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of ear-level device 210A and 210B is a device structured for right ear attachment, while the other is a device structured for left ear attachment.

[0046] BTE device 610 includes a behind-the-earmodule (BTE module) 611 and an in-the-ear module (ITE module) 612. ITE module 612 includes a custom or modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0047] FIG. 7 is a block diagram showing one embodiment of the circuit of ear-level device shown in FIG. 6 used as ear-level device 110. The exterior configuration of ear-level device 110 is of the type illustrated as BTE device 610.

[0048] Ear-level device 110 configured as BTE device 610 differs from ear-level device 110 configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 7 illustrates one embodiment in which BTE module 611 includes external microphone 420, processor 460, wireless transceiver 450, antenna 455, VOX circuit 430, and battery 480, and ITE module 612 includes ear canal microphone 440 and ear canal speaker 470. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, behind-ear portion 611 includes only battery 480 to maximize the duration of use of ear-level device 110 between battery replacements or recharges.

[0049] FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as each of ear-level devices 210A and 210B. The exterior configuration of each of ear-level devices 210A and 210B is of the type illustrated as BTE device 610.

[0050] Ear-level devices 210A/210B configured as BTE device 610 differ from ear-level devices 210A/210B configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 8 illustrates one embodiment of ear-level device 210A including a BTE module 611A and an ITE module 612A, and ear-level device 210B including a BTE module 611B and an ITE module 612B. BTE module 611A includes processor 560A, wireless transmitter 550A, antenna 555A, transmitter VOX circuit 530A, and battery 580A. ITE module 612A includes ear canal microphone 540. Behind-the-ear portion 611B includes external microphone 520, processor 560B, wireless receiver 550B, antenna 555B, receiver VOX circuit 530B, and battery 580B. ITE module 612B includes ear canal speaker 570. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, BTE modules 611A and 611B include only batteries 580A and 580B, respectively, to maximize the duration of use of ear-level devices 210A and 210B, respectively, between battery replacements or recharges.

[0051] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication using system 100 or system 200. The method provides for private, simultaneous two-way communication between a person and a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. In this embodiment. VOX is used for power management. A VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 or system 200 that are voice activated. The VOX is "on" when the voice-activated portions of the circuit are activated, i.e., when the power switch remains on. It is "started" by turning on the power switch, and "stopped" by turning off the power switch.

[0052] Steps 910-918 illustrate the process of detecting a sound from an ear canal and transmitting it to a remote device. A microphone disposed in an ear canal detects an occluded sound within the ear canal at 910. If the occluded sound is detected at 911, the VOX of the system is started at 912 if it is not already on. The occluded sound is enhanced at 914 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice of a speech, and the noise is any sound other than the voice of the speech. The occluded sound is then converted to an outgoing radio signal at 916. The outgoing radio signal representing the occluded sound is transmitted to the remote device at 918.

[0053] Steps 920-928 illustrate the process of detecting a sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects an incoming radio signal representing a remote sound from the remote device at 920. If the incoming radio signal is detected at 921, the VOX of the system is started at 922 if it is not already on. The incoming radio signal is converted to a first incoming audio signal (the remote sound) at 924. The first incoming audio signal is enhanced at 926 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice from the remote device, and the noise is any sound other than the voice.

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A speaker disposed in the ear canal transmits the first incoming audio signal representing the sound from the remote device to the ear canal at 928.

[0054] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928.

[0055] If no signal (sound or radio signal) is detected at 911, 921, and 931, the VOX is stopped at 940. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times. In one embodiment, the VOX is stopped after a predetermined period of time to avoid frequent activation/deactivation during a conversation.

[0056] Systems 100 and 200 are each a full duplex system capable of performing steps 910-918, 920-928, and 930-938 are performed in parallel and simultaneously. In an embodiment in which system 100 performs the method, ear-level device 110 performs steps 910-918, 924-928, and 930-938. If a signal is detected at 911, 921, or 931, VOX 430 activates at least portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively. If no signal is detected at 911, 921, or 931, VOX 430 deactivates at least the portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively.

[0057] In an embodiment in which system 200 performs the method, ear-level device 210A performs steps 910-918, and ear-level device 210B performs steps 920-928 and 930-930. If a signal is detected at 911, transmitter VOX 530A activates at least portions of device 210A performing steps 914-918. If a signal is detected at 921 or 931, receiver VOX 530B activates at least portions of device 210B performing steps 924-928 or 934-938, respectively. If no signal is detected at 911, transmitter VOX 530A deactivates at least the portions of device 210A performing steps 914-918. If no signal is detected at 921 or 931, receiver VOX 530B deactivates at least the portions of device 210B performing steps 924-928 or 934-938, respectively.

[0058] FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication using system 100. In this embodiment, in addition to power management, the VOX is also used for gating, i.e., blanking or attenuating, sounds. The VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 that are voice activated and gating modules capable of blanking or attenuating the occluded sound, the remote sound, and the ambient sound. Examples of such gating modules include occluded sound gating module 441, remote sound gating module 451, and ambient sound gating module 421, designated for the occluded sound, the remote sound, and the ambient sound, respectively.

[0059] Steps 1010-1019 illustrate the process of detecting the occluded sound from the ear canal and transmitting it to the remote device. The microphone disposed in the ear canal detects the occluded sound within the ear canal at 1010. If the occluded sound is detected at 1011, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1012 if they are not already powered. The VOX determines whether to start gating the occluded sound on at 1013. In one embodiment, the occluded sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. Examples of gating coordination rules are described above with respect to FIG. 4B. If the occluded sound is gated on at 1013, it is processed at 1014 by following steps that are identical or similar to steps 914-916-918. The occluded sound is gated off at 1019 if it has been gated on but is no longer detected. In one embodiment, the occluded sound is gated off after it is not detected for a predetermined period.

[0060] Steps 1020-1028 illustrate the process of detecting the remote sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects the incoming radio signal representing the remote sound from the remote device at 1020. If the remote sound is detected at 1021, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1022 if they are not already powered. The VOX determines whether to start gating the remote sound on at 1023. In one embodiment, the remote sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. In another embodiment, the remote sound is gated on when the incoming radio signal is detected and when the effective gating coordination rules permits. If the remote sound is gated on at 1023, it is processed at 1024 by following steps that are identical or similar to steps 924-926-928. The remote sound is gated off at 1029 if it has been gated on but is no longer detected. In one embodiment, the remote sound is gated off after it is not detected for a predetermined period.

[0061] Steps 1030-1038 illustrate the process of detecting the ambient sound and transmit it to the ear canal. A microphone outside the ear canal detects the ambient sound at 1030. If the ambient sound is detected at 1031, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1032 if they are not already powered. The VOX determines whether to start gating the ambient sound on at 1033. In one embodiment, the ambient sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. If the

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ambient sound is gated on at 1033, it is processed at 1034 by following steps that are identical or similar to steps 934-936-938. The ambient sound is gated off at 1039 if it has been gated on but is no longer detected. In one embodiment, the ambient sound is gated off after it is not detected for a predetermined period.

[0062] If no signal (sound or radio signal) is detected at 1011,1021, and 1031, the VOX is stopped at 1040. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times, In one embodiment, the VOX is stopped after a predetermined period of time to avoid Sequent activation/deactivation during a conversation.

[0063] It is to be understood that the above detailed description is intended to be illustrative, and not restrictive. Other embodiments will be apparent to those of skill in the art upon reading and understanding the above description. For example, two ear-level devices 110, or a pair of devices each being similar to ear-level device 110, can be used to deliver a stereo sound. The scope of the invention should, therefore, be determined with reference to the appended claims.

Claims

1. A system adapted for use in at least one of a first ear (101A) having a first ear canal (303) and a second ear (101B) having a second ear canal (303) comprising:

   a first ear-level device (110, 210A) configured to be one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device, the first ear-level device including:

   a first microphone (440) adapted for use in one of the first ear canal and the second ear canal, and adapted to receive an occluded sound (305) within the one of the first ear canal (303) and the second ear canal (303); and
   a wireless receiver, connected to a first processor (460), to receive a first wireless signal representing a remote sound;

   wherein the first processor (460) is connected to the first microphone (440) and is adapted to process the occluded sound (305) and the remote sound; the first ear level device further including:

   a speaker adapted for use in the one of the first ear canal (303) and the second ear canal (303) and adapted to transmit the processed remote sound to the one of the first ear canal (303) and the second ear canal (303);
   a wireless transmitter, connected to the first processor (460) and adapted to receive the processed occluded sound from the first processor (460) and transmit a second wireless signal representing the occluded sound;
   a first earmold housing at least the first microphone (440) and the speaker (470), at least a portion of the first earmold configured for placement in the one of the first ear canal (303) and the second ear canal (303); and
   a first voice operated exchange (VOX) circuit (430) connected to the first processor (460);
   the first VOX circuit (430) including:

   an occluded sound gating module (441) adapted to gate the occluded sound on, and to gate the remote sound off, when the occluded sound is detected while the remote sound is not detected; and
   a remote sound gating module (451) adapted to gate the remote sound on, and to gate the occluded sound off, when the incoming remote sound is detected while the occluded sound is not detected.

2. A system according to claim 1, wherein the first earmold houses the first microphone (440), the first processor (460) and the wireless transmitter, and wherein the first earmold is configured for use as the ITE device (310A).

3. A system according to claim 1, wherein the first ear level device (110,210A) comprises an ITE module including the first microphone (440) and a BTE module attached to the ITE module, the BTE module including the first processor and the wireless transmitter.

4. A system according to claim 1, wherein the first microphone (440) and the speaker (470) are implemented as one common device receiving the occluded sound and transmitting the processed remote sound.

5. A system according to claim 4, further comprising a second microphone (420) connected to the first processor (460), the second microphone (420) adapted to receive an ambient sound from outside the one of the first ear canal (303) and the second ear canal (303), and wherein:

   the first processor (460) further processes the ambient sound; and
   the speaker (470) is configured to transmit the processed remote and ambient sounds to the one of the first ear canal (303) and the second ear canal (303).

6. A system according to claim 5, wherein the first VOX

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circuit (430) comprises an ambient sound gating module (421), coupled to the first microphone (420), to blank or substantially attenuate the ambient sound.

7. A system according to any of the preceding claims, wherein the first processor (460) comprises a speech recognition module.

8. A system according to any of claims 4 - 7, wherein the first earmold houses the first microphone (440), the first processor (460), the wireless transmitter, the wireless receiver, and the speaker (470), and wherein the first earmold is configured for use as the ITE device.

9. A system according to claim 8, wherein the first earmold is configured for use as an in-the-canal (ITC) device (310B).

10. A system according to claim 9, wherein the first earmold is configured for use as a completely-in-the-canal (CIC) device (310C).

11. A system according to any of claims 4 - 7, wherein the first ear level device (110,210A) comprises an ITE module including the first microphone (440) and the speaker (470) and a BTE module attached to the ITE module and including the second microphone (420), the first processor (460), the wireless transmitter and the wireless receiver.

12. A system according to claim 11, wherein the BTE module comprises the first voice operated exchange (VOX) circuit (430).

13. A system according to claim 11, wherein the BTE module comprises a rechargeable battery (480).

14. A system according to any of claims 7 - 13, further comprising a remote device (190, 290) communicatively coupled to the wireless transmitter and the wireless receiver via a telemetry link (195, 295) providing for simultaneous bidirectional communication between the first ear-level device (110, 210A) and the remote device (190, 290).

15. A system according to claim 14, wherein the remote device (190, 290) comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

16. A system according to claim 1, wherein the first microphone (440) is adapted for use in the first ear canal (303) and receives the occluded sound within the first ear canal (303), wherein the at least the portion of the first earmold is configured for placement in the first ear canal (303), and further comprising a second ear-level device (210B) including:

a wireless receiver to receive a first wireless signal representing a remote sound;
a second processor (260), connected to the wireless receiver, to convert the first wireless signal to the remote sound;
a speaker (470) connected to the second processor, the speaker adapted for use in the second ear canal (303) to transmit the remote sound to the second ear canal (303); and
a second earmold housing at least the speaker, at least a portion of the second earmold configured for placement in the second ear canal (303).

17. A system according to claim 16, wherein the second ear-level device (210B) further comprises a second microphone (420) connected to the second processor (460), the second microphone (420) adapted to receive an ambient sound from outside the second ear canal (303), and wherein:

the second processor (460) further processes the ambient sound; and
the speaker (470) is configured to transmit the remote sound and the processed ambient sound to the second ear canal (303).

18. A system according to any of claims 16 and 17, wherein the first ear-level device (110,210A) comprises the first voice operated exchange (VOX) circuit (430) connected to the first processor (460), and the second ear-level device (210B) further comprises a second VOX circuit (430) connected to the second processor.

19. A system according to any of claims 16 - 18, wherein the first processor and the second processor each comprise a speech recognition module.

20. A system according to any of claims 16 - 19, wherein the first and second ear-level devices are each the BTE device.

21. A system according to any of claims 16 - 19, wherein the first and second ear-level devices are each the ITE device.

22. A system according to claim 21, wherein the first and second ear-level devices are each an in-the-canal (ITC) device (310B).

23. A system according to claim 22, wherein the first and second ear-level devices are each a completely-in-the-canal (CIC) device (310C).

24. A system according to any of claims 16 - 23, further

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comprising a remote device (190,290) communicatively coupled to the first and second ear-level devices.

25. A system according to claim 24, wherein the remote device (190, 290) comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

26. A method, comprising:

detecting an occluded sound using a first microphone adapted for use in an ear canal (303);
detecting an incoming radio signal representing a remote sound from a remote device (190, 290);
starting a voice operated exchange (VOX) housed within an ear-level communication device when at least one of the occluded sound and the incoming radio signal is detected if the VOX is not already on, the ear level communication device being one of an in-the-ear (ITE) device and a behind-the-ear (BTE) device;
gating the occluded sound on, and gating the remote sound off, when the occluded sound is detected while the incoming radio signal is not detected; and
gating the remote sound on, and gating the occluded sound off, when the incoming radio signal is detected while the occluded sound is not detected.

27. A method according to claim 26, wherein detecting the occluded sound comprises comparing an amplitude of the occluded sound to an occluded sound threshold, wherein the occluded sound is detected when the amplitude of the occluded sound exceeds the occluded sound threshold.

28. A method according to any of claims 26 and 27, wherein detecting the incoming radio signal comprises comparing an amplitude of the remote sound to a remote sound threshold, wherein the incoming radio signal is detected when the amplitude of the remote sound exceeds the remote sound threshold.

29. A method according to any of claims 26 - 28, further comprising detecting an ambient sound using a second microphone, and wherein starting the VOX comprises starting the VOX when at least one of the occluded sound, the incoming radio signal, and the ambient sound is detected if the VOX is not already on.

30. A method according to claim 29, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound off, when the incoming radio signal is detected while the occluded sound is not detected.

31. A method according to claim 29, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound on, when the incoming radio signal is detected while the occluded sound is not detected.

32. A method according to claim 29, further comprising gating the occluded sound on, gating the remote sound off, and gating the ambient sound off, when the occluded sound is detected while the incoming radio signal is not detected.

Patentansprüche

1. System, das zur Verwendung in einem ersten Ohr (101A) mit einem ersten Gehörgang (303) und/oder einem zweiten Ohr (101B) mit einem zweiten Gehörgang (303) eingerichtet ist, mit:

einer ersten ohrebenen- bzw. ohrseitigen Vorrichtung (110, 210A), die dafür eingerichtet ist, eine In-Ohr-Vorrichtung (ITE) oder eine Hinter-Ohr-Vorrichtung (BTE) zu sein, wobei die erste ohrseitige Vorrichtung beinhaltet:

ein erstes Mikrofon (440), das zur Verwendung in dem ersten Gehörgang oder dem zweiten Gehörgang eingerichtet ist, und das dafür eingerichtet ist, einen verdeckten Ton (305) in jeweils dem ersten Gehörgang (303) oder dem zweiten Gehörgang (303) zu empfangen; und
einen Drahtlosempfänger, der mit einem ersten Prozessor (460) verbunden ist, um ein erstes Drahtlossignal zu empfangen, welches einen Fernton darstellt;

wobei der erste Prozessor (460) mit dem ersten Mikrofon (440) verbunden und dafür eingerichtet ist, den verdeckten Ton (305) und den Fernton zu verarbeiten;
wobei die erste ohrseitige Vorrichtung ferner beinhaltet:

einen Lautsprecher, der zur Verwendung in dem jeweiligen ersten Gehörgang (303) oder dem zweiten Gehörgang (303) eingerichtet ist, und der dafür eingerichtet ist, den verarbeiteten Fernton an den jeweiligen ersten Gehörgang (303) oder den zweiten Gehörgang (303) zu übertragen;
einen Drahtlossender, der mit dem ersten Prozessor (460) verbunden und dafür eingerichtet ist, den verarbeiteten verdeckten Ton von dem ersten Prozessor (460) zu empfangen und ein zweites Drahtlossignal, das den verdeckten Ton darstellt, zu senden;

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ein erstes Ohrformgehäuse wenigstens bei dem ersten Mikrofon (440) und dem Lautsprecher (470), wobei wenigstens ein Teil der ersten Ohrform dafür eingerichtet ist, in dem ersten Gehörgang (303) oder dem zweiten Gehörgang (303) platziert zu werden; und
eine erste stimmbetriebene Austauschschaltung (VOX) (430), die mit dem ersten Prozessor (460) verbunden ist;

wobei die erste VOX-Schaltung (430) beinhaltet:

ein Ausblendmodul (441) für einen verdeckten Ton, das dafür eingerichtet ist, den verdeckten Ton einzublenden und den Fernton auszublenden, wenn der verdeckte Ton bestimmt wird, während der Fernton nicht bestimmt wird; und
ein Ausblendmodul (441) für einen Fernton, das dafür eingerichtet ist, den Fernton einzublenden und den verdeckten Ton auszublenden, wenn der eingehende Fernton bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

2. System nach Anspruch 1, bei dem eine erste Ohrform das erste Mikrofon (440), den ersten Prozessor (460) und den Drahtlossender einschließt, und bei dem die erste Ohrform zur Verwendung als ITE-Vorrichtung (310) eingerichtet ist.

3. System nach Anspruch 1, bei dem die erste ohrseitige Vorrichtung (110, 210A) ein ITE-Modul, das das erste Mikrofon (440) beinhaltet, und ein zweites BTE-Modul, das an das ITE-Modul angebracht ist, beinhaltet, wobei das ITE-Modul den ersten Prozessor und den Drahtlossender beinhaltet.

4. System nach Anspruch 1, bei dem das erste Mikrofon (440) und der Lautsprecher (470) als eine gemeinsame Vorrichtung implementiert sind, die den verdeckten Ton empfängt und den verarbeiteten Fernton sendet.

5. System nach Anspruch 4, ferner mit einem zweiten Mikrofon (420), das mit dem ersten Prozessor (460) verbunden ist, wobei das zweite Mikrofon (420) dafür eingerichtet ist, einen ümgebungston von außerhalb das jeweiligen ersten Gehörgangs (303) oder des zweiten Gehörgangs (303) zu empfangen, und wobei:

der erste Prozessor (460) den Umgebungston ferner bzw. weiterverarbeitet; und
der Lautsprecher (470) dafür eingerichtet ist, den verarbeiteten Fern- und Umgebungston an den jeweiligen ersten Gehörgang (303) und zweiten Gehörgang (303) zu senden.

6. System nach Anspruch 5, bei dem die erste VOX-Schaltung (430) ein Umgebungstonausblendmodul (421) aufweist, das mit dem ersten Mikrofon (420) verbunden ist, um den Umgebungston auf Null zu setzen oder ihn wesentlich zu dämpfen.

7. System nach einem der vorstehenden Ansprüche, bei dem der erste Prozessor (460) ein Spracherkennungsmodul aufweist.

8. System nach einem der Ansprüche 4 bis 7, bei dem die erste Ohrform, das erste Mikrofon (440), den ersten Prozessor (460), den Drahtlossender, den Drahtlosempfänger und den Lautsprecher (470) einschließt, und bei dem die erste Ohrform dafür eingerichtet ist, als die ITE-Vorrichtung verwendet zu werden.

9. System nach Anspruch 8, bei dem die erste Ohrform zur Verwendung als In-Kanal-Vorrichtung (ITC) (310B) verwendet zu werden.

10. System nach Anspruch 9, bei dem die erste Ohrform zur Verwendung als vollständig-In-Kanal-Vorrichtung (CIC) (310C) verwendet zu werden.

11. System nach einem der Ansprüche 4 bis 7, bei dem die erste ohrseitige Vorrichtung (110, 210A) ein ITE-Modul aufweist, das das erste Mikrofon (440) und den Lautsprecher (470) und ein BTE-Modul, das an das ITE-Modul angebracht ist, und das das zweite Mikrofon (420), den ersten Prozessor (460), den drahtlos Sender und den drahtlos Empfänger beinhaltet, beinhaltet.

12. System nach Anspruch 11, bei dem das BTE-Modul die erste sprachbetriebene Austauschschaltung (VOX) (340) aufweist.

13. System nach Anspruch 11, bei dem das BTE-Modul eine widerauffladbare Batterie (480) aufweist.

14. System nach einem der Ansprüche 7 bis 13, das ferner eine entfernte Vorrichtung (190, 290) aufweist, die mit dem Drahtlossender und dem Drahtlosempfänger über eine Telemetrieverbindung (195, 295) in Kommunikationsverbindung steht, um eine gleichzeitige bi-direktionale Kommunikation zwischen der ersten ohrseitigen Vorrichtung (110, 210A) und der entfernten Vorrichtung (190, 290) bereitzustellen.

15. System nach Anspruch 14, bei dem die entfernte Vorrichtung (190, 290) wenigstens einen Computer, einen Personal Digital Assistant (PDA), ein Mobiltelefon, ein Walkie Talkie und/oder einen Sprachübersetzer aufweist.

16. System nach Anspruch 1, bei dem das erste Mikro-

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fon (440) dafür eingerichtet ist, in dem ersten Gehörgang (303) verwendet zu werden, und dieses den verdeckten Ton in dem ersten Gehörgang (303) empfängt, wobei der wenigstens eine Abschnitt der ersten Ohrform zur Anordnung in dem ersten Gehörgang (303) eingerichtet ist, und ferner mit einer zweiten ohrseitigen Vorrichtung (210B), die beinhaltet:

einen Drahtlosempfänger, um ein erstes Drahtlossignal zu empfangen, das einen Fernton darstellt;
einen zweiten Prozessor (260), der mit dem Drahtlosempfänger verbunden ist, um das erste Drahtlossignal in den Fernton umzuwandeln;
einen Lautsprecher (470), der mit dem zweiten Prozessor verbunden ist, wobei der Lautsprecher zur Verwendung in dem zweiten Gehörgang (303) eingerichtet ist, um den Fernton an den zweiten Gehörgang (303) zu übertragen; und
einer zweiten Ohrform, die wenigstens den Lautsprecher einschließt, wobei wenigstens ein Teil der zweiten Ohrform zur Anordnung in dem zweiten Gehörgang (303) eingerichtet ist.

17. System nach Anspruch 16, bei dem die zweite ohrseitige Vorrichtung (210B) ferner ein zweites Mikrofon (420) aufweist, das mit dem zweiten Prozessor (460) verbunden ist, wobei das zweite Mikrofon (420) dafür eingerichtet ist, einen Umgebungston von außerhalb des zweiten Gehörgangs (303) zu empfangen, und bei dem:

der zweite Prozessor (460) den Umgebungston weiterverarbeitet; und
der Lautsprecher (470) dafür eingerichtet ist, den Fernton und den verarbeiteten Umgebungston an den zweiten Gehörgang (303) zu senden.

18. System nach einem der Ansprüche 16 oder 17, bei dem die erste ohrseitige Vorrichtung (110, 210A) die erste sprachbetriebene Austauschschaltung (VOX) (430) mit dem ersten Prozessor (460) verbunden aufweist, und wobei die zweite ohrseitige Vorrichtung (210B) ferner eine zweite VOX-Schaltung (430) aufweist, die mit dem zweiten Prozessor verbunden ist.

19. System nach einem der Ansprüche 16 bis 18, bei dem der erste Prozessor und der zweite Prozessor jeweils ein Spracherkennungsmodul aufweisen.

20. System nach einem der Anspruch 16 bis 19, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils die BTE-vorrichtungen darstellen.

21. System nach einem der Ansprüche 16 bis 19, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils die ITE-Vorrichtungen darstellen.

22. System nach Anspruch 21, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils eine In-Gehörgangs-Vorrichtung (ITC) (310B) darstellen.

23. System nach Anspruch 22, bei dem die ersten und zweiten ohrseitigen Vorrichtungen jeweils Vollständig-In-Gehörgangs-Vorrichtungen (CIC) (310C) darstellen.

24. System nach einem der Ansprüche 16 bis 23, das ferner eine entfernte Vorrichtung (190, 290) aufweist, die kommunikativ mit den ersten und zweiten ohrseitigen Vorrichtungen verbunden ist.

25. System nach Anspruch 24, bei dem die entfernte Vorrichtung (190, 290) wenigstens einen Computer, einen Personal Digital Assistant (PDA), ein Mobiltelefon, ein Walkie Talkie und/oder einen Sprachübersetzer aufweist.

26. Verfahren, das beinhaltet:

Bestimmen eines verdeckten Tons unter Verwendung eines ersten Mikrofons, das zur Verwendung in einem Gehörgang (303) eingerichtet ist;
Bestimmen eines eingehenden Funksignals, das einen Fernton von einer entfernten Vorrichtung (190, 290) darstellt;
Starten einer sprachbetriebenen Austauscheinrichtung (VOX), die in einer ohrseitigen Kommunikationsvorrichtung eingeschlossen ist, wenn der verdeckte Ton und/oder das eingehende Funksignal bestimmt wird, wenn die VOX nicht bereits an ist, wobei die ohrseitige Kommunikationsvorrichtung eine In-Ohr-Vorrichtung (ITE) oder eine Hinter-Ohr-Vorrichtung (BTE) darstellt;
Einblenden des verdeckten Tons und Ausblenden des Ferntons, wenn der verdeckte Ton bestimmt wird, während das eingehende Funksignal nicht bestimmt wird; und
Einblenden des Ferntons und Ausblenden des verdeckten Tons, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

27. Verfahren nach Anspruch 26, bei dem das Bestimmen des verdeckten Tons, das Vergleichen einer Amplitude des verdeckten Tons mit einem Schwellwert für einen verdeckten Ton beinhaltet, wobei der verdeckte Ton bestimmt wird, wenn die Amplitude des verdeckten Tones den Schwellwert für den verdeckten Ton überschreitet.

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28. Verfahren nach einem der Ansprüche 26 oder 27, bei dem das Bestimmen des eingehenden Funksignals das Vergleichen einer Amplitude des Ferntons mit einem Ferntonschwellwert beinhaltet, wobei das eingehende Funksignal bestimmt wird, wenn die Amplitude des Ferntons den Ferntonschwellwert überschreitet.

29. Verfahren nach einem der Ansprüche 26 bis 28, das ferner das Bestimmen eines Umgebungstons unter Verwendung eines zweiten Mikrofons beinhaltet, und wobei das Starten der VOX das Starten der VOX beinhaltet, wenn wenigstens der verdeckte Ton, das eingehende Funksignal und/oder der Umgebungston bestimmt wird, während die VOX noch nicht bereits an ist.

30. Verfahren nach Anspruch 29, das ferner das Ausblenden des verdeckten Tons, das Einblenden des Ferntons und das Ausblenden des Umgebungstons beinhaltet, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

31. Verfahren nach Anspruch 29, das ferner das Ausblenden des verdeckten Tons, das Einblenden des Ferntons und das Einblenden des Umgebungstons beinhaltet, wenn das eingehende Funksignal bestimmt wird, während der verdeckte Ton nicht bestimmt wird.

32. Verfahren nach Anspruch 29, das ferner das Einblenden des verdeckten Tones, das Ausblenden des Ferntons und das Ausblenden des Umgebungstons beinhaltet, wenn der verdeckte Ton bestimmt wird, während das eingehende Funksignal nicht bestimmt wird.

Revendications

1. Un système adapté en vue d'une utilisation dans au moins une parmi une première oreille (101A) ayant un premier conduit auditif (303) et une seconde oreille (101B) ayant un second conduit auditif (303), comprenant:

un premier dispositif d'oreille (110, 210A) configuré pour être un parmi un dispositif placé à l'intérieur de l'oreille (ITE) et un dispositif contour d'oreille (BTE), le premier dispositif d'oreille comprenant :

un premier microphone (440) adapté en vue d'une utilisation dans un parmi le premier conduit auditif et le second conduit auditif, et adapté pour recevoir un son occlus (305) à l'intérieur de celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; et
un récepteur sans fil, connecté à un premier processeur (460), pour recevoir un premier signal sans fil représentant un son distant ;

dans lequel le premier processeur (460) est connecté au premier microphone (440) et est adapté pour traiter le son occlus (305) et le son distant; le premier dispositif d'oreille comprenant en outre :

un haut-parleur adapté en vue d'une utilisation dans celui parmi le premier conduit auditif (303) et le second conduit auditif (303) et adapté pour transmettre le son distant traité à celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; un émetteur sans fil, connecté au premier processeur (460) et adapté pour recevoir le son occlus traité du premier processeur (460) et émettre un second signal sans fil représentant le son occlus ;
un premier embout auriculaire logeant au moins le premier microphone (440) et le haut-parleur (470), au moins une portion du premier embout auriculaire étant configurée en vue d'être placée dans celui parmi le premier conduit auditif (303) et le second conduit auditif (303) ; et
un premier circuit d'échange à commande vocale (VOX) (430) connecté au premier processeur (460) ;
le premier circuit VOX (430) comprenant :

un module de contrôle de son occlus (441) adapté pour laisser passer le son occlus et pour bloquer le son distant lorsque le son occlus est détecté alors que le son distant n'est pas détecté ; et
un module de contrôle de son distant (451) adapté pour laisser passer le son distant et pour bloquer le son occlus lorsque le son distant entrant est détecté alors que le son occlus n'est pas détecté.

2. Système selon la revendication 1, dans lequel le premier embout auriculaire loge le premier microphone (440), le premier processeur et l'émetteur sans fil, et dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif ITE (310 A).

3. Système selon la revendications 1, dans lequel le premier dispositif d'oreille (110, 210A) comprend un module ITE comprenant le premier microphone (440) et un module BTE fixé au module ITE, le module BTE comprenant le premier processeur et l'émetteur sans fil.

4. Système selon la revendication 1, dans lequel le premier microphone (440) et le haut-parleur (470) sont

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mis en oeuvre sous la forme d'un dispositif commun recevant le son occlus et émettant le son distant traité.

5. Système selon la revendication 4, comprenant en outre un second microphone (420) connecté au premier processeur (460), le second microphone (420) étant adapté pour recevoir un son ambiant de l'extérieur de celui parmi le premier conduit auditif (303) et le second conduit auditif (303), et dans lequel :

le premier processeur (460) traite en outre le son ambiant ; et
le haut-parleur (470) est configuré pour transmettre les sons distant et ambiant traités à celui parmi le premier conduit auditif (303) et le second conduit auditif (303).

6. Système selon la revendication 5, dans lequel le premier circuit VOX (430) comprend un module de contrôle de son ambiant (421), couplé au premier microphone (420), pour occulter ou atténuer sensiblement le son ambiant.

7. Système selon l'une quelconque des revendications précédentes, dans lequel le premier processeur (460) comprend un module de reconnaissance vocale.

8. Système selon l'une quelconque des revendications 4 à 7, dans lequel le premier embout auriculaire loge le premier microphone (440), le premier processeur (460), l'émetteur sans fil, le récepteur sans fil, et le haut-parleur (470), et dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif ITE.

9. Système selon la revendication 8, dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif intra-conduit (ITC) (310B).

10. Système selon la revendication 9, dans lequel le premier embout auriculaire est configuré pour être utilisé comme dispositif complètement intra-conduit (CIC) (310C).

11. Système selon l'une quelconque des revendications 4 à 7, dans lequel le premier dispositif d'oreille (110, 210A) comprend un module ITE comprenant le premier microphone (440) et le haut-parleur (470) et un module BTE fixé au module ITE et comprenant le second microphone (420), le premier processeur (460), l'émetteur sans fil et le récepteur sans fil.

12. Système selon la revendication 11, dans lequel le module BTE comprend le premier circuit d'échange à commande vocale (VOX) (430).

13. Système selon la revendication 11, dans lequel le module BTE comprend une batterie rechargeable (480).

14. Système selon l'une quelconque des revendications 7 à 13, comprenant en outre un dispositif distant (190, 290) couplé en communication à l'émetteur sans fil et au récepteur sans fil via une liaison de télémesure (195, 295) assurant une communication bidirectionnelle simultanée entre le premier dispositif d'oreille (110, 210A) et le dispositif distant (190, 290).

15. Système selon la revendication 14, dans lequel le dispositif distant (190, 290) comprend au moins un parmi un ordinateur, un assistant numérique personnel (PDA), un téléphone cellulaire, un talkie-walkie, ou un traducteur de langue.

16. Système selon la revendication 1, dans lequel le premier microphone (440) est adapté en vue d'une utilisation dans le premier conduit auditif (303) et reçoit le son occlus à l'intérieur du premier conduit auditif (303), ans lequel au moins la portion du premier embout auriculaire est configurée en vue d'être placée dans le premier conduit auditif (303), et comprenant en outre un second dispositif d'oreille (210B) comprenant :

un récepteur sans fil pour recevoir un premier signal sans fil représentant un son distant ;
un second processeur (260), connecté au récepteur sans fil, pour convertir le premier signal sans fil dans le son distant ;
un haut-parleur (470) connecté au second processeur, le haut-parleur étant adapté en vue d'une utilisation dans le second conduit auditif (303) pour transmettre le son distant au second conduit auditif (303) ; et
un second embout auriculaire logeant au moins le haut-parleur, au moins une portion du second embout auriculaire étant configurée en vue d'être placée dans le second conduit auditif (303).

17. Système selon la revendication 16, dans lequel le second dispositif d'oreille (210B) comprend en outre un second microphone (420) connecté au second processeur (460), le second microphone (420) étant adapté pour recevoir un son ambiant de l'extérieur du second conduit auditif (303), et dans lequel :

le second processeur (460) traite en outre le son ambiant ; et le haut-parleur (470) est configuré pour transmettre le son distant et le son ambiant traité au second conduit auditif (303).

18. Système selon l'une quelconque des revendications 16 et 17, dans lequel le premier dispositif d'oreille

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(110, 210) comprend le premier circuit d'échange à commande vocale (VOX) (430) connecté au premier processeur (460), et le second dispositif d'oreille (210B) comprend en outre un second circuit VOX (430) connecté au second processeur.

19. Système selon l'une quelconque des revendications 16 à 18, dans lequel le premier processeur et le second processeur comprennent chacun un module de reconnaissance vocale.

20. Système selon l'une quelconque des revendications 16 à 19, dans lequel les premier et second dispositifs d'oreille sont respectivement le dispositif BTE.

21. Système selon l'une quelconque des revendications 16 à 19, dans lequel les premier et second dispositifs d'oreille sont respectivement le dispositif ITE.

22. Système selon la revendication dans lequel les premier et second dispositifs d'oreille sont respectivement un dispositif intra-conduit (ITC) (310B).

23. Système selon la revendication 22, dans lequel les premier et second dispositifs d'oreille sont respectivement un dispositif complètement intra-conduit (CIC) (310C).

24. Système selon l'une quelconque des revendications 16 à 23, comprenant en outre un dispositif distant (190, 290) couplé en communication aux premier et second dispositifs d'oreille.

25. Système selon la revendication 24, dans lequel le dispositif distant (190, 290) comprend au moins un parmi un ordinateur, un assistant numérique personnel (PDA), un téléphone cellulaire, un talkie walkie, ou un traducteur de langue.

26. Un procédé comprenant les étapes consistant à :

détecter un son occlus au moyen d'un premier microphone adapté en vue d'une utilisation dans un conduit auditif (303) ;
détecter un signal radio entrant représentant un son distant provenant d'un dispositif distant (190, 290) ;
lancer un échange à commande vocale (VOX) logé à l'intérieur d'un dispositif de communication d'oreille lorsqu'au moins un parmi le son occlus et le signal radio entrant est détecté si le VOX n'est pas déjà en marche, le dispositif de communication d'oreille étant un parmi un dispositif placé à l'intérieur de l'oreille (ITE) et un dispositif contour d'oreille (BTE) ;
laisser passer le son occlus, et bloquer le son distant lorsque le son distant occlus est détecté alors que le signal radio entrant n'est pas détecté ; et
laisser passer le son distant, et bloquer le son occlus lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

27. Procédé selon la revendication 26, dans lequel la détection du son occlus comprend la comparaison d'une amplitude du son occlus avec un seuil de son occlus, dans lequel le son occlus est détecté lorsque l'amplitude du son occlus dépasse le seuil du son occlus.

28. Procédé selon l'une quelconque des revendications 26 et 27, dans lequel la détection du signal radio entrant comprend la comparaison d'une amplitude du son distant avec un seuil de son distant, dans lequel le signal radio entrant est détecté lorsque l'amplitude du son distant dépasse le seuil du son distant.

29. Procédé selon l'une quelconque des revendications 26 à 28, comprenant en outre la détection d'un son ambiant au moyen d'un second microphone, et dans lequel le lancement du VOX comprend le lancement du VOX lorsqu'au moins un parmi le son occlus, le signal radio entrant et le son ambiant est détecté si le VOX n'est pas déjà en marche.

30. Procédé selon la revendications 29, comprenant en outre l'étape consistant à bloquer le son occlus, laisser passer le son distant et bloquer le son ambiant lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

31. Procédé selon la revendication 29, comprenant en outre l'étape consistant à bloquer le son occlus, laisser passer le son distant et laisser passer le son ambiant lorsque le signal radio entrant est détecté alors que le son occlus n'est pas détecté.

32. Procédé selon la revendication 29, comprenant en outre l'étape consistant à laisser passer le son occlus, bloquer le son distant, et bloquer le son ambiant lorsque le son occlus est détecté alors que le signal radio entrant n'est pas détecté.

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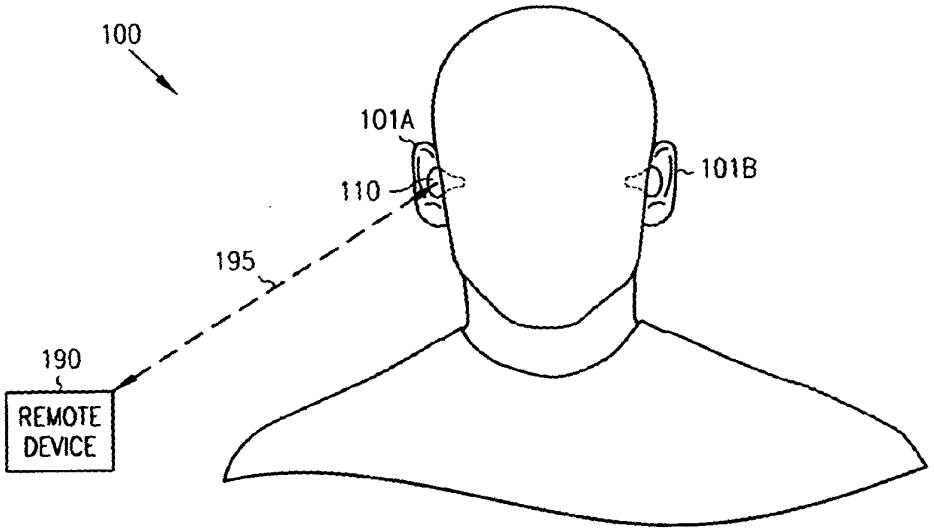

100
101A
101B
110
195
190
REMOTE DEVICE

FIG. 1

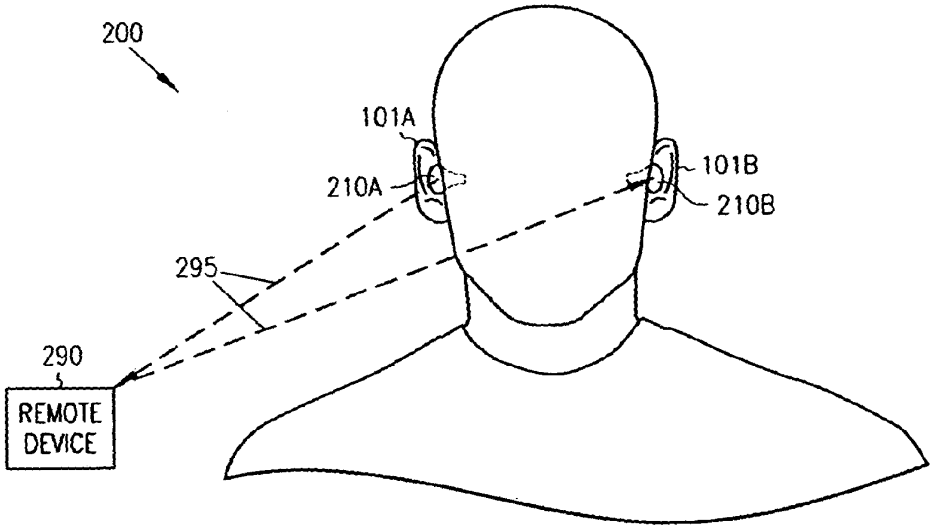

200
101A
101B
210A
210B
295
290
REMOTE DEVICE

FIG. 2

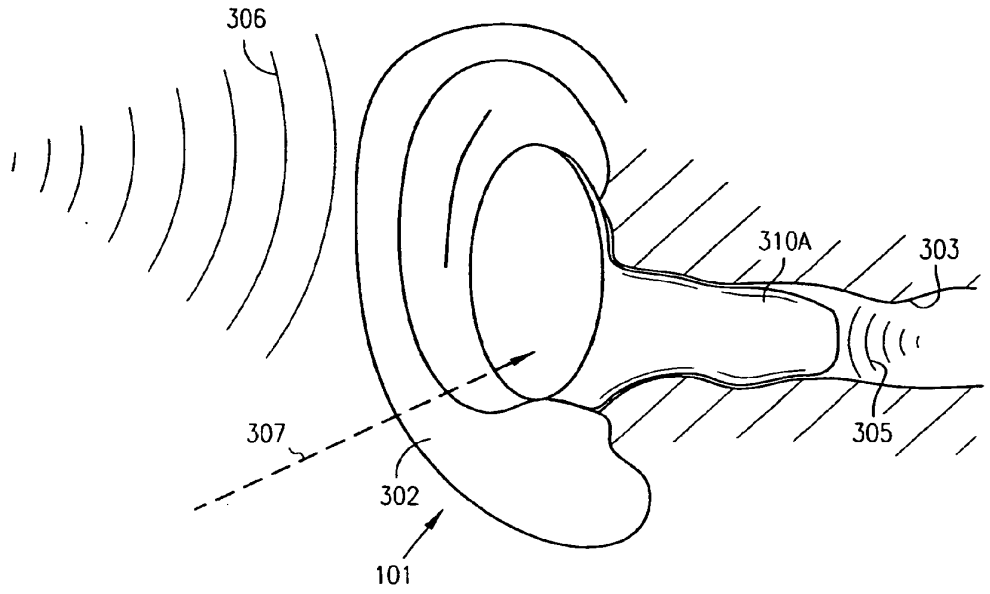


FIG. 3A

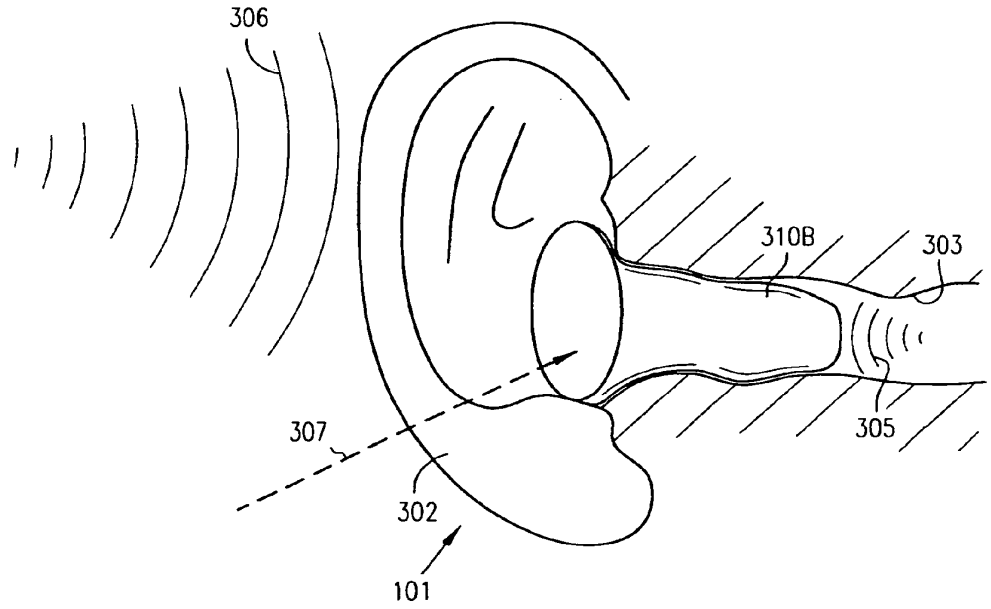


FIG. 3B

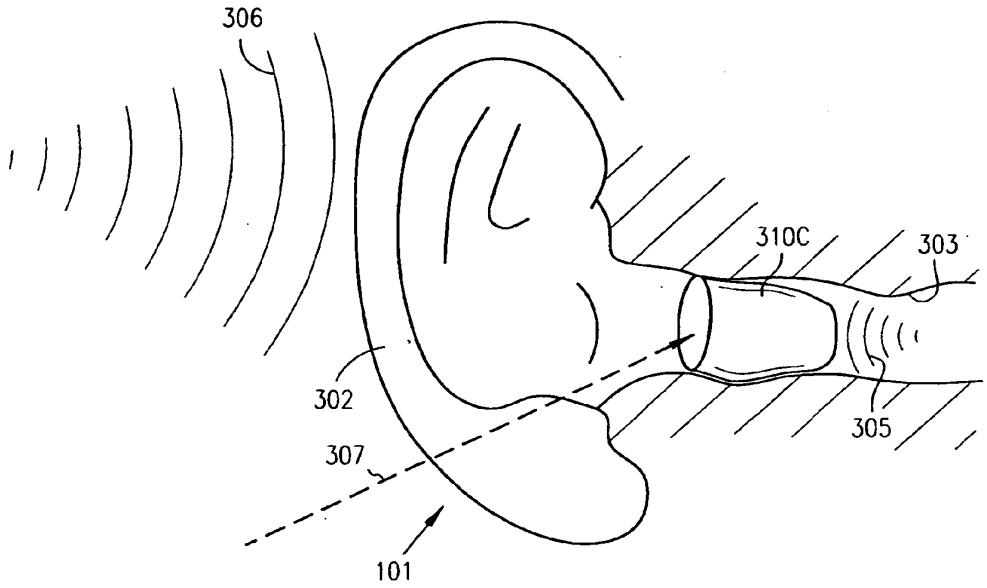


FIG. 3C

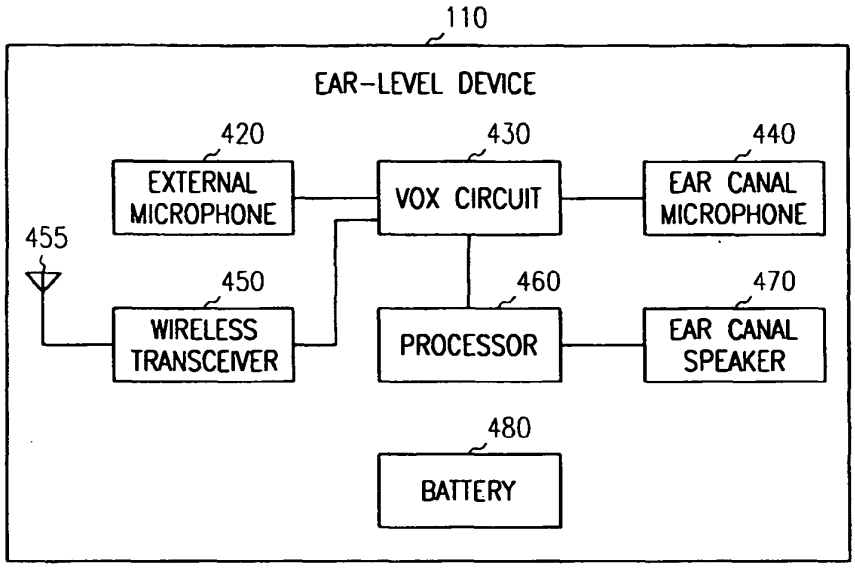

110
EAR-LEVEL DEVICE
420
EXTERNAL MICROPHONE
430
VOX CIRCUIT
440
EAR CANAL MICROPHONE
455
450
WIRELESS TRANSCEIVER
460
PROCESSOR
470
EAR CANAL SPEAKER
480
BATTERY

FIG. 4A

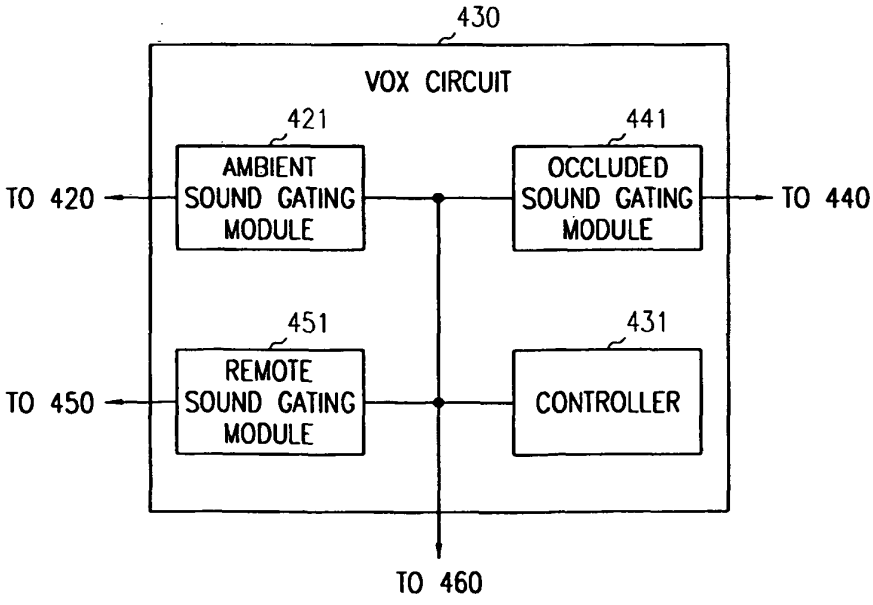

430
VOX CIRCUIT
421
AMBIENT SOUND GATING MODULE
441
OCCLUDED SOUND GATING MODULE
TO 420
TO 440
451
REMOTE SOUND GATING MODULE
431
CONTROLLER
TO 450
TO 460

FIG. 4B

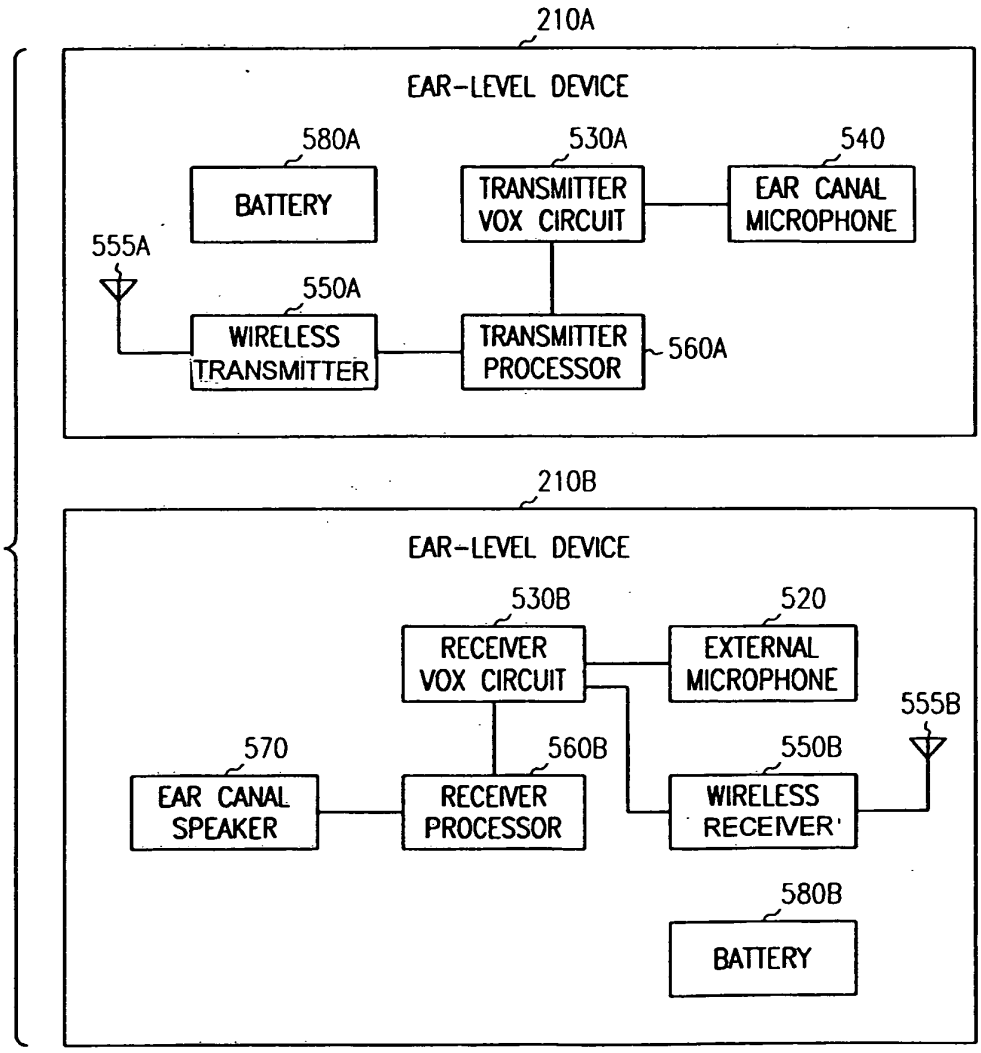

210A
EAR-LEVEL DEVICE
580A
BATTERY
530A
TRANSMITTER VOX CIRCUIT
540
EAR CANAL MICROPHONE
555A
550A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
560A
210B
EAR-LEVEL DEVICE
530B
RECEIVER VOX CIRCUIT
520
EXTERNAL MICROPHONE
555B
570
EAR CANAL SPEAKER
560B
RECEIVER PROCESSOR
550B
WIRELESS RECEIVER
580B
BATTERY

FIG. 5

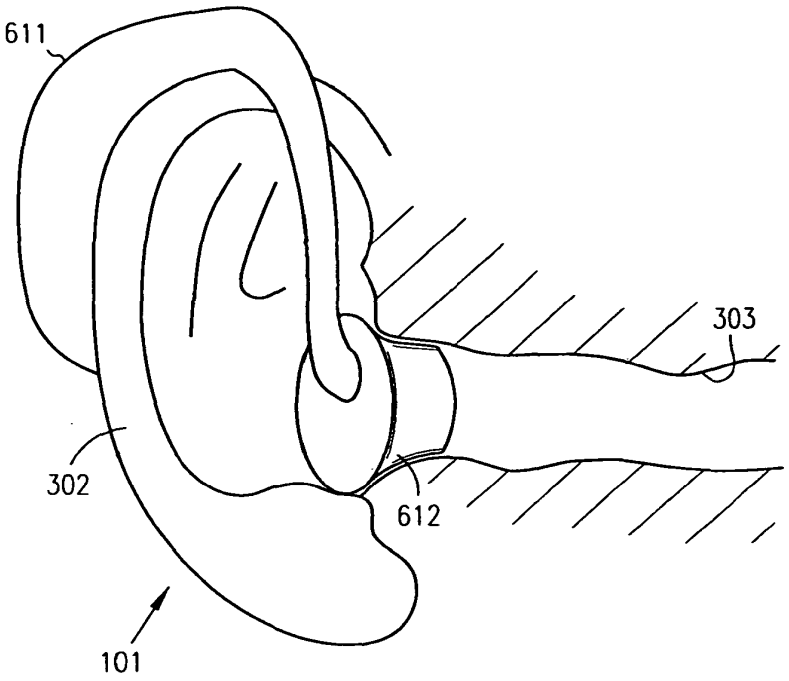

611
303
302
612
101

FIG. 6

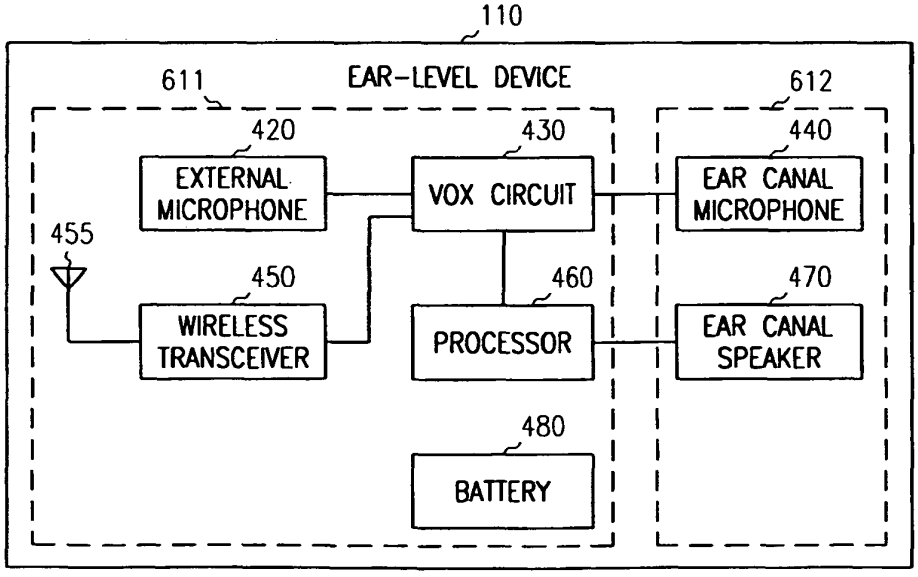

110
611
EAR-LEVEL DEVICE
612
420
EXTERNAL MICROPHONE
430
VOX CIRCUIT
440
EAR CANAL MICROPHONE
455
450
WIRELESS TRANSCEIVER
460
PROCESSOR
470
EAR CANAL SPEAKER
480
BATTERY

FIG. 7

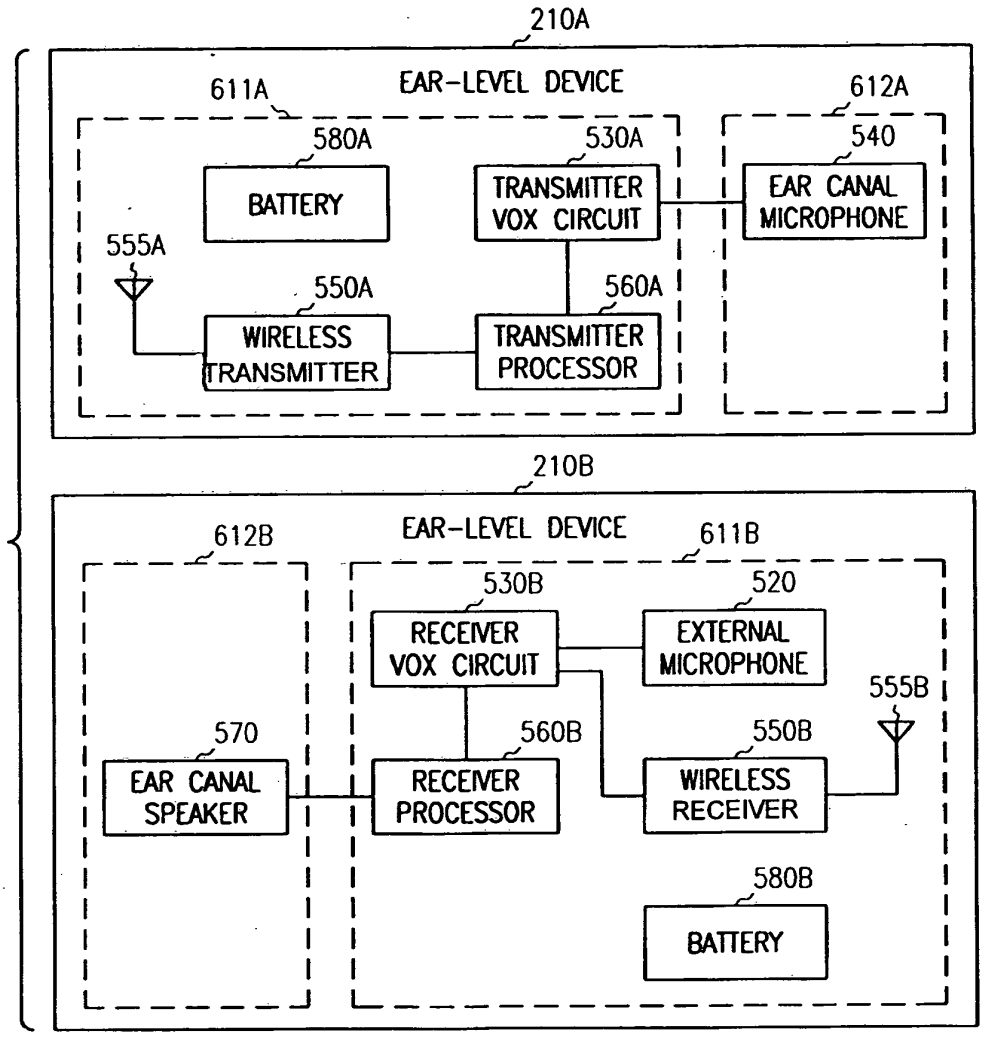

210A
EAR-LEVEL DEVICE
611A
612A
580A
530A
540
BATTERY
TRANSMITTER VOX CIRCUIT
EAR CANAL MICROPHONE
555A
550A
560A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
210B
EAR-LEVEL DEVICE
612B
611B
530B
520
RECEIVER VOX CIRCUIT
EXTERNAL MICROPHONE
555B
570
560B
550B
EAR CANAL SPEAKER
RECEIVER PROCESSOR
WIRELESS RECEIVER
580B
BATTERY

FIG. 8

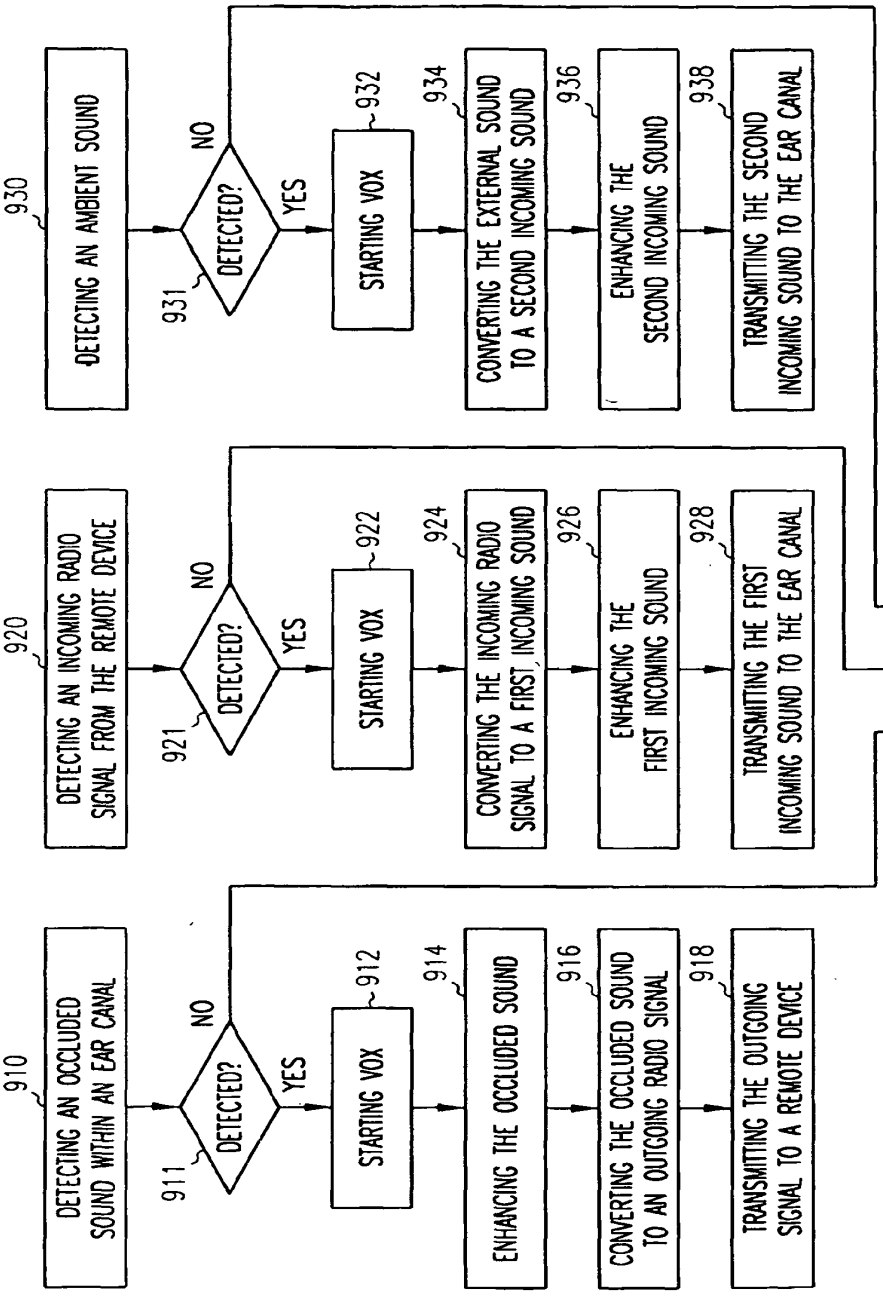

910
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
911
DETECTED?
YES
NO
912
STARTING VOX
914
ENHANCING THE OCCLUDED SOUND
916
CONVERTING THE OCCLUDED SOUND TO AN OUTGOING RADIO SIGNAL
918
TRANSMITTING THE OUTGOING SIGNAL TO A REMOTE DEVICE
920
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
921
DETECTED?
YES
NO
922
STARTING VOX
924
CONVERTING THE INCOMING RADIO SIGNAL TO A FIRST INCOMING SOUND
926
ENHANCING THE FIRST INCOMING SOUND
928
TRANSMITTING THE FIRST INCOMING SOUND TO THE EAR CANAL
930
DETECTING AN AMBIENT SOUND
931
DETECTED?
YES
NO
932
STARTING VOX
934
CONVERTING THE EXTERNAL SOUND TO A SECOND INCOMING SOUND
936
ENHANCING THE SECOND INCOMING SOUND
938
TRANSMITTING THE SECOND INCOMING SOUND TO THE EAR CANAL
940
STOPPING VOX

FIG. 9

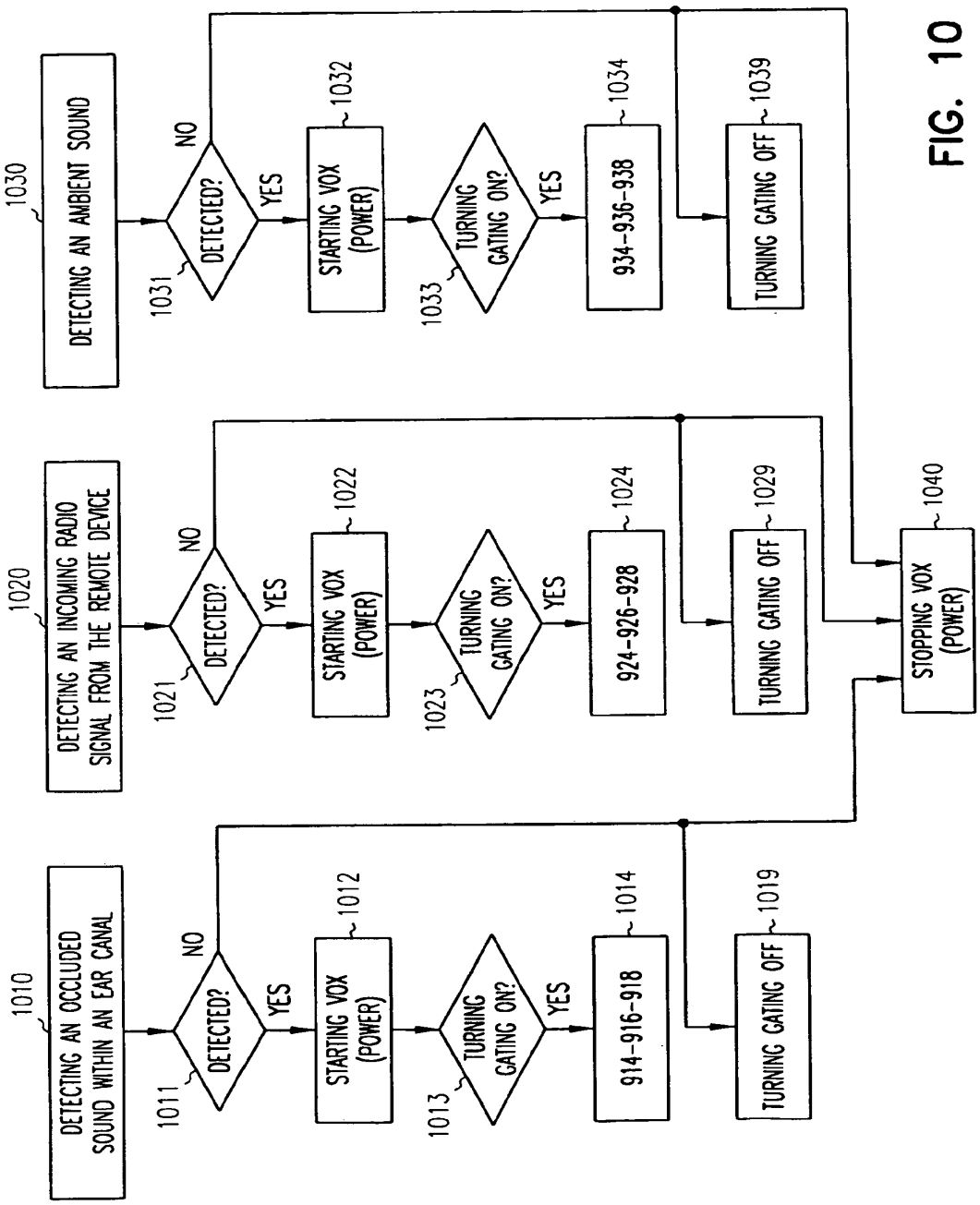

1010
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
1011
DETECTED?
NO
YES
1012
STARTING VOX (POWER)
1013
TURNING GATING ON?
YES
1014
914-916-918
1019
TURNING GATING OFF
1020
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
1021
DETECTED?
NO
YES
1022
STARTING VOX (POWER)
1023
TURNING GATING ON?
YES
1024
924-926-928
1029
TURNING GATING OFF
1030
DETECTING AN AMBIENT SOUND
1031
DETECTED?
NO
YES
1032
STARTING VOX (POWER)
1033
TURNING GATING ON?
YES
1034
934-936-938
1039
TURNING GATING OFF
1040
STOPPING VOX (POWER)

FIG. 10

REFERENCES CITED IN THE DESCRIPTION

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- WO 03073790 A **[0010]**

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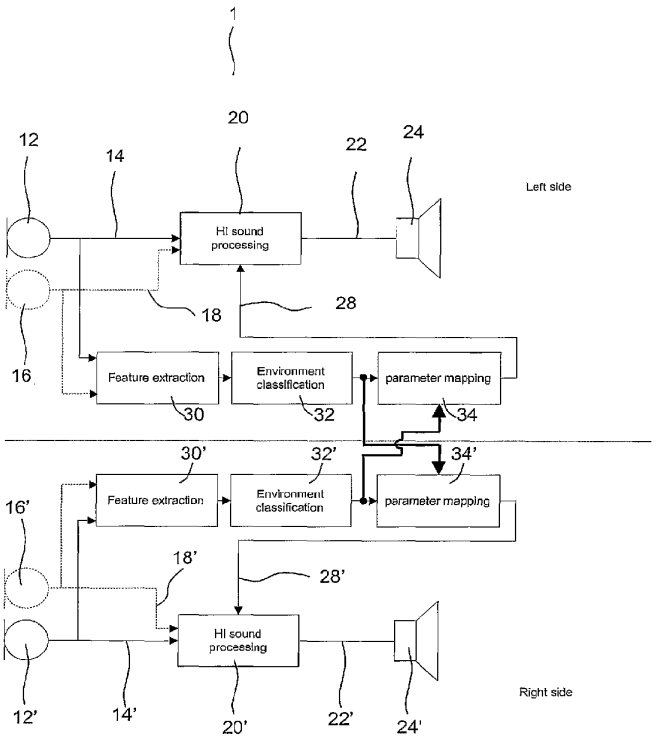
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[Continued on next page]

(54) Title: A BINAURAL HEARING AID SYSTEM WITH COORDINATED SOUND PROCESSING



(57) Abstract: The present invention relates to a binaural hearing aid system comprising a first hearing aid and a second hearing aid, each of which comprises a microphone and an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment, a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal, and a binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid for provision of outputs for each of the first and second hearing aids for selection of the signal processing algorithm of each of the respective hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

WO 2004/114722 A1

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— *with international search report*

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A BINAURAL HEARING AID SYSTEM WITH COORDINATED SOUND PROCESSING

FIELD OF THE INVENTION

The present invention relates to a binaural hearing aid system with a first hearing aid and a second hearing aid, each of which comprises a microphone, an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment, a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal.

BACKGROUND OF THE INVENTION

Today's conventional hearing aids typically comprise a Digital Signal Processor (DSP) for processing of sound received by the hearing aid for compensation of the user's hearing loss. As is well known in the art, the processing of the DSP is controlled by a signal processing algorithm having various parameters for adjustment of the actual signal processing performed. The gains in each of the frequency channels of a multi-channel hearing aid are examples of such parameters.

The flexibility of the DSP is often utilized to provide a plurality of different algorithms and/or a plurality of sets of parameters of a specific algorithm. For example, various algorithms may be provided for noise suppression, i.e. attenuation of undesired signals and amplification of desired signals. Desired signals are usually speech or music, and undesired signals can be background speech, restaurant clatter, music (when speech is the desired signal), traffic noise, etc.

The different algorithms or parameter sets are typically included to provide comfortable and intelligible reproduced sound quality in different sound environments, such as speech, babble speech, restaurant clatter, music, traffic noise, etc. Audio signals obtained from different sound environments may possess very different characteristics, e.g. average and maximum sound pressure levels (SPLs) and/or frequency content. Therefore, in a hearing aid with a DSP, each type of sound environment may be associated with a particular program wherein a particular setting of algorithm parameters of a signal processing algorithm provides processed sound of optimum signal quality in a specific sound environment. A set of such parameters may typically include parameters related to broadband gain, corner frequencies or slopes of frequency-selective filter algorithms and

parameters controlling e.g. knee-points and compression ratios of Automatic Gain Control (AGC) algorithms.

Consequently, today's DSP based hearing instruments are usually provided with a number of different programs, each program tailored to a particular sound environment category and/or particular user preferences. Signal processing characteristics of each of these programs is typically determined during an initial fitting session in a dispenser's office and programmed into the instrument by activating corresponding algorithms and algorithm parameters in a non-volatile memory area of the hearing aid and/or transmitting corresponding algorithms and algorithm parameters to the non-volatile memory area.

Some known hearing aids are capable of automatically classifying the user's sound environment into one of a number of relevant or typical everyday sound environment categories, such as speech, babble speech, restaurant clatter, music, traffic noise, etc.

Obtained classification results may be utilised in the hearing aid to automatically select signal processing characteristics of the hearing aid, e.g. to automatically switch to the most suitable algorithm for the environment in question. Such a hearing aid will be able to maintain optimum sound quality and/or speech intelligibility for the individual hearing aid user in various sound environments.

US 5,687,241 discloses a multi-channel DSP based hearing instrument that utilises continuous determination or calculation of one or several percentile values of input signal amplitude distributions to discriminate between speech and noise input signals. Gain values in each of a number of frequency channels are adjusted in response to detected levels of speech and noise.

However, it is often desirable to provide a more subtle characterization of a sound environment than only discriminating between speech and noise. As an example, it may be desirable to switch between an omni-directional and a directional microphone preset program in dependence of, not just the level of background noise, but also on further signal characteristics of this background noise. In situations where the user of the hearing aid communicates with another individual in the presence of the background noise, it would be beneficial to be able to identify and classify the type of background noise. Omni-directional operation could be selected in the event that the noise being traffic noise to allow the user to clearly hear approaching traffic independent of its direction of arrival. If, on the other hand, the background noise was classified as being babble-noise, the directional listening program could be selected to allow the user to hear a target speech signal with improved signal-to-noise ratio (SNR) during a conversation.

Applying Hidden Markov Models for analysis and classification of the microphone signal may obtain a detailed characterisation of e.g. a microphone signal. Hidden Markov Models are capable of modelling stochastic and non-stationary signals in terms of both short and long time temporal variations. Hidden Markov Models have been applied in speech recognition as a tool for modelling statistical properties of speech signals. The article "A Tutorial on Hidden Markov Models and Selected Applications in Speech Recognition", published in Proceedings of the IEEE, VOL 77, No. 2, February 1989 contains a comprehensive description of the application of Hidden Markov Models to problems in speech recognition.

WO 01/76321 discloses a hearing aid that provides automatic identification or classification of a sound environment by applying one or several predetermined Hidden Markov Models to process acoustic signals obtained from the listening environment. The hearing aid may utilise determined classification results to control parameter values of a signal processing algorithm or to control switching between different algorithms so as to optimally adapt the signal processing of the hearing aid to a given sound environment.

The different available signal processing algorithms may change the signal characteristics significantly. In binaural hearing aid systems, it is therefore important that the determination of sound environment does not differ for the two hearing aids. However, since sound characteristics may differ significantly at the two ears of a user, it will often occur that sound environment determination at the two ears of a user differs, and this leads to undesired different signal processing of sounds for each of the ears of the user.

SUMMARY OF THE INVENTION

Thus, there is a need for a binaural hearing aid system wherein sound environment determination does not differ for the two hearing aids so that signal processing in the two hearing aids may be coordinated and the user be provided with desired processed sound in both ears at the same time.

According to the present invention, this and other objects are solved by provision of a binaural hearing aid system of the above-mentioned type wherein the hearing aids are connected either by wire or by a wireless link to at least one binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid whereby the sound environment is determined and classified based on binaural signals. The one or more binaural sound environment detectors provide outputs for each of the first and second hearing aids for selection of the

signal processing algorithm of each of the hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

In this way both hearing aids may process sound in response to a common determination of sound environment. Sound environment determination may be performed by one common environment detector, for example situated in one of the hearing aids or in a remote control, or, by a plurality of environment detectors, such as an environment detector in each of the first and second hearing aids.

In the event that the user has substantially the same hearing loss in both ears and the sound environment is omni-directional, i.e. the sound environment does not change with direction, coordination of sound processing in the hearing aids leads to execution of identical signal processing algorithms in the respective signal processors of the hearing aids. In the event that the hearing aid user suffers from a binaural hearing loss, the signal processing algorithms may desirably differ for compensation of the different binaural hearing losses.

It is an important advantage of the present invention that binaural sound environment detection is more accurate than monaural detection since signals from both ears are taken into account.

It is a further advantage of the present invention that signal processing in the hearing aids of the binaural hearing aid system is coordinated since the sound environment detection is the same for both hearing aids.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the present invention reference will now be made, by way of example, to the accompanying drawings, in which:

Fig. 1 illustrates schematically a prior art monaural hearing aid with sound environment classification,

Fig. 2 illustrates schematically a first embodiment of the present invention,

Fig. 3 illustrates schematically a second embodiment of the present invention,

Fig. 4 illustrates schematically a third embodiment of the present invention, and

Fig. 5 illustrates schematically a fourth embodiment of the present invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Fig. 1 illustrates schematically a prior art monaural hearing aid 10 with sound environment classification.

The monaural hearing aid 10 comprises a first microphone 12 and a first A/D converter (not shown) for provision of a digital input signal 14 in response to sound signals received at the microphone 12 in a sound environment, and a second microphone 16 and a second A/D converter (not shown) for provision of a digital input signal 18 in response to sound signals received at the microphone 16, a processor 20 that is adapted to process the digital input signals 14, 18 in accordance with a predetermined signal processing algorithm to generate a processed output signal 22, and a D/A converter (not shown) and an output transducer 24 for conversion of the respective processed sound signal 22 to an acoustic output signal.

The hearing aid 10 further comprises a sound environment detector 26 for determination of the sound environment surrounding a user of the hearing aid 10. The determination is based on the output signals of the microphones 12, 16. Based on the determination, the sound environment detector 26 provides outputs 28 to the hearing aid processor 20 for selection of the signal processing algorithm appropriate in the determined sound environment. Thus, the hearing aid processor 20 is automatically switched to the most suitable algorithm for the determined environment whereby optimum sound quality and/or speech intelligibility is maintained in various sound environments.

The signal processing algorithms of the processor 20 may perform various forms of noise reduction and dynamic range compression as well as a range of other signal processing tasks.

The sound environment detector 26 comprises a feature extractor 30 for determination of characteristic parameters of the received sound signals. The feature extractor 30 maps the unprocessed sound inputs 14, 18 sound features, i.e. the characteristic parameters. These features can be signal power, spectral data and other well-known features.

The sound environment detector 26 further comprises an environment classifier 32 for categorizing the sound environment based on the determined characteristic parameters. The environment classifier categorizes the sounds into a number of environmental classes, such as speech, babble speech, restaurant clatter, music, traffic noise, etc. The classification process may consist of a simple nearest neighbour search, a neural network, a Hidden Markov Model system or another system capable of pattern recognition. The output of the environmental classification can be a "hard" classification containing one

single environmental class or a set of probabilities indicating the probabilities of the sound belonging to the respective classes. Other outputs may also be applicable.

The sound environment detector 26 further comprises a parameter map 34 for the provision of outputs 28 for selection of the signal processing algorithms.

The parameter map 34 maps the output of the environment classification 32 to a set of parameters for the hearing aid sound processor 20. Examples of such parameters are amount of noise reduction, amount of gain and amount of HF gain. Other parameters may be included.

Figs. 2-5 illustrate various preferred embodiments of the present invention. The illustrated binaural hearing aid system 1 comprises a first hearing aid 10 and a second hearing aid 10', each of which comprises a first microphone 12, 12 ' and an A/D converter (not shown) and a second microphone 16, 16' and A/D converter (not shown) for provision of a digital input signals 14, 14', 18, 18' in response to sound signals received at the respective microphones 12, 12', 16, 16' in a sound environment, a processor 20, 20' that is adapted to process the digital input signals 14, 18, 14', 18' in accordance with a predetermined signal processing algorithm to generate a processed output signal 22, 22', and a D/A converter (not shown) and an output transducer 24, 24' for conversion of the respective processed sound signals 22, 22' to an acoustic output signal

In the embodiments illustrated in Figs. 2-4, each of the hearing aids 10, 10' of the binaural hearing aid system 1 further comprises a binaural sound environment detector 26, 26' for determination of the sound environment surrounding a user of the binaural hearing aid system 1. The determination is based on the output signals of the microphones 12, 12', 16, 16'. Based on the determination, the binaural sound environment detector 26, 26' provides outputs 28, 28' to the hearing aid processors 20, 20' for selection of the signal processing algorithm appropriate in the determined sound environment. Thus, the binaural sound environment detectors 26, 26' determines the sound environment based on signals from both hearing aids, i.e. binaurally, whereby hearing aid processors 20, 20' is automatically switched in co-ordination to the most suitable algorithm for the determined environment whereby optimum sound quality and/or speech intelligibility is maintained in various sound environments by the binaural hearing aid system 1.

The binaural sound environment detectors 26, 26' illustrated in Figs. 2-4 are both similar to the binaural sound environment detector shown in Fig. 1 apart from the fact that the monaural environment detector only receives inputs from one hearing aid while each of the binaural sound environment detectors 26, 26' receives inputs from both hearing aids. Thus,

according to the present invention, signals are transmitted between the hearing aids 10, 10' so that the algorithms executed by the signal processors 20, 20' are selected in co-ordination, e.g. in case of an omni-directional sound environment, i.e. the sound environment does not change with direction, the algorithms are selected to be identical apart from possible differences in hearing loss compensation of the two ears.

In the embodiment of Fig. 2, the unprocessed signals 14, 14', 18, 18' from the microphones 12, 12', 16, 16' of one hearing aid 10, 10' are transmitted to the other hearing aid and inputted to the respective feature extractor 30, 30'. Thus, feature extraction in each of the hearing aids is based on the identical four input signals so that identical sound environment characteristic parameters will be determined binaurally in both hearing aids 10, 10'.

The signals may be transmitted in analogue form or in digital form, and the communication channel may be wired or wireless.

In the embodiment shown in Fig. 3, the output 36, 36' of the feature extractor 30, 30' of one hearing aid 10, 10' is transmitted to the respective other hearing aid 10', 10. The environment classifier 32, 32' then operates on two sets of features 36, 36' to determine the environment. Since both environment classifiers 32, 32' receive the same data, they will produce the same output.

In the embodiment shown in Fig. 4, the output 38, 38' of the environment classifier 32, 32' of one hearing aid 10, 10' is transmitted to the respective other hearing aid 10, 10'. The parameter map 34, 34' then operates on two inputs 38, 38' to produce the parameters for the processor algorithms, but since both parameter mapping units 34, 34' receive identical inputs, identical parameter values will be produced.

This embodiment has a number of advantages: Usually classification systems take both past and present data into account – they have memory. This makes them sensitive to missing data, since a classification requires a complete data set. Therefore it is required that the data link is safe, in the sense that data is guaranteed to be transmitted. The parameter mapping can be implemented without memory so that only present data is taken into account when generating parameters. This makes the system robust to packet loss and latency, since the parameter mapping may simply re-use old data in the event that data is missing. This will of course delay the correct action, but to the user the systems will appear to be synchronized.

The transmission data rate is low, since only a set of probabilities or logic values for the environment classes has to be transmitted.

Rather high latency can be accepted. By applying time constants to the variables that will change according to the output of the parameter mapping it is possible to smooth out any differences that is caused by latency. As described earlier it is important that signal processing in the two hearing instruments is coordinated. However if transition periods of a few seconds are allowed the system can operate with only 3-4 transmissions per second. Hereby, power consumption is kept low.

A binaural hearing aid system 1 with a remote control 40 is shown in Fig. 5. The environment detector 26 is positioned in the remote control 40. The required signals are transmitted to and from both hearing aids.

CLAIMS

1. A binaural hearing aid system comprising

a first hearing aid and a second hearing aid, each of which comprises

a microphone and an A/D converter for provision of a digital input signal in response to sound signals received at the respective microphone in a sound environment,

a processor that is adapted to process the digital input signals in accordance with a predetermined signal processing algorithm to generate a processed output signal, and

a D/A converter and an output transducer for conversion of the respective processed sound signal to an acoustic output signal, and

a binaural sound environment detector for binaural determination of the sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid for provision of outputs for each of the first and second hearing aids for selection of the signal processing algorithm of each of the respective hearing aid processors so that the hearing aids of the binaural hearing aid system perform coordinated sound processing.

2. A system according to claim 1, wherein the binaural sound environment detector having inputs for each of the digital input signals is provided in a remote control of the system.

3. A system according to claim 1, wherein the binaural sound environment detector is provided in one of the hearing aids and provides an output for the other hearing aid.

4. A system according to claim 1, wherein each of the hearing aids comprises a binaural sound environment detector.

5. A system according to any of the preceding claims, wherein the binaural sound environment detector comprises

a feature extractor for determination of characteristic parameters of the received sound signals,

an environment classifier for categorizing the sound environment based on the determined characteristic parameters, and

a parameter map for the provision of outputs for selection of the signal processing algorithms.

6. A system according to claim 4 and 5, wherein each of the feature extractors has inputs for each of the digital input signals.

7. A system according to claim 4 and 5, wherein each of the environment classifiers has inputs connected to each of the feature extractors.

8. A system according to claim 4 and 5, wherein each of the parameter maps has inputs for connected to each of the environment classifiers.

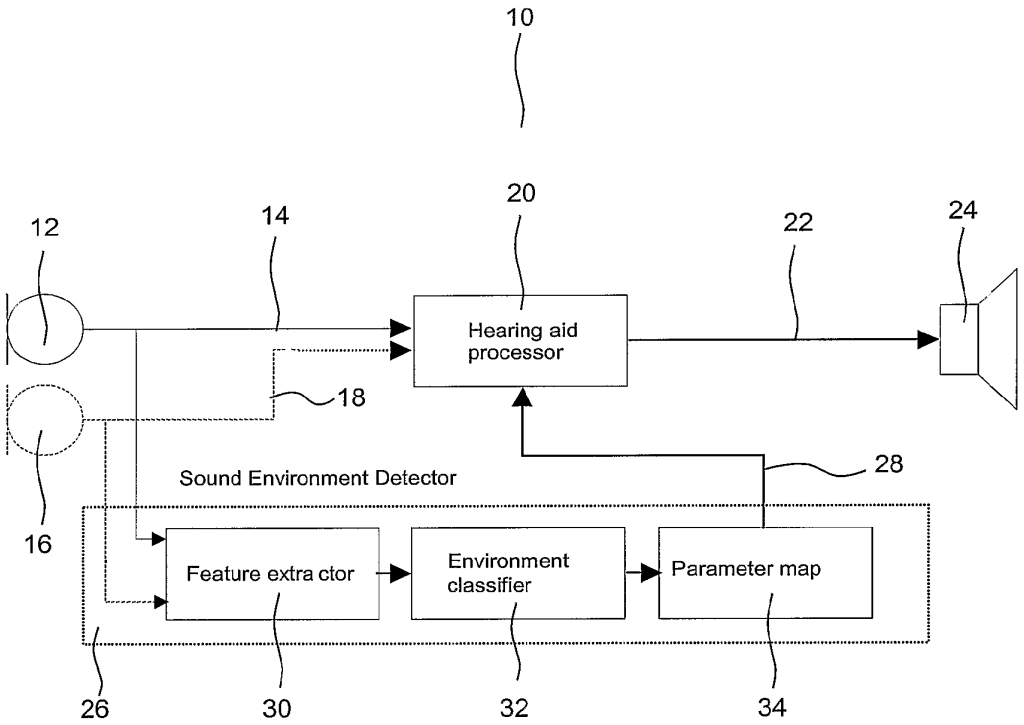

10
12
14
20
22
24
Hearing aid processor
18
16
Sound Environment Detector
28
Feature extra ctor
Environment classifier
Parameter map
26
30
32
34

Fig. 1

2/5

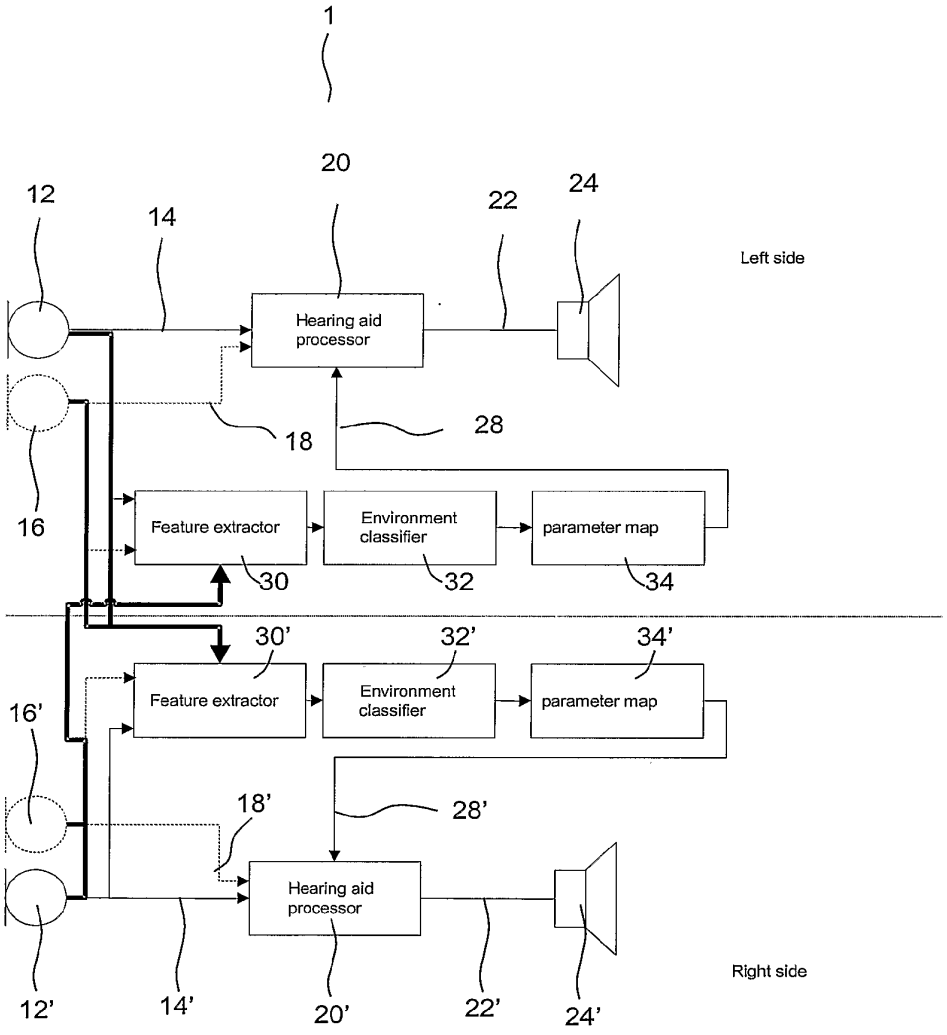

1
20
12
14
22
24
Left side
Hearing aid processor
18
28
16
Feature extractor
Environment classifier
parameter map
30
32
34
30'
32'
34'
Feature extractor
Environment classifier
parameter map
16'
18'
28'
Hearing aid processor
12'
14'
20'
22'
24'
Right side

Fig. 2

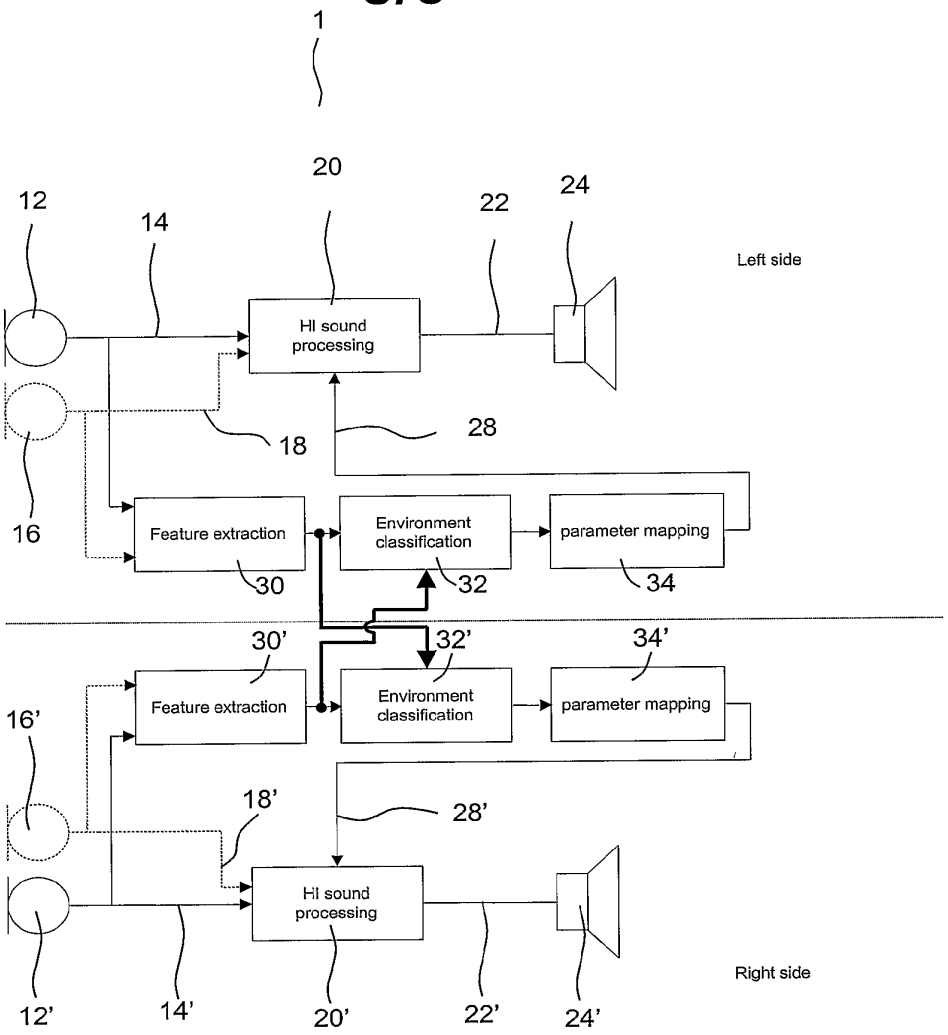

1
20
12
14
22
24
Left side
HI sound processing
18
28
16
Feature extraction
Environment classification
parameter mapping
30
32
34
30'
32'
34'
16'
Feature extraction
Environment classification
parameter mapping
18'
28'
HI sound processing
Right side
12'
14'
20'
22'
24'

Fig. 3

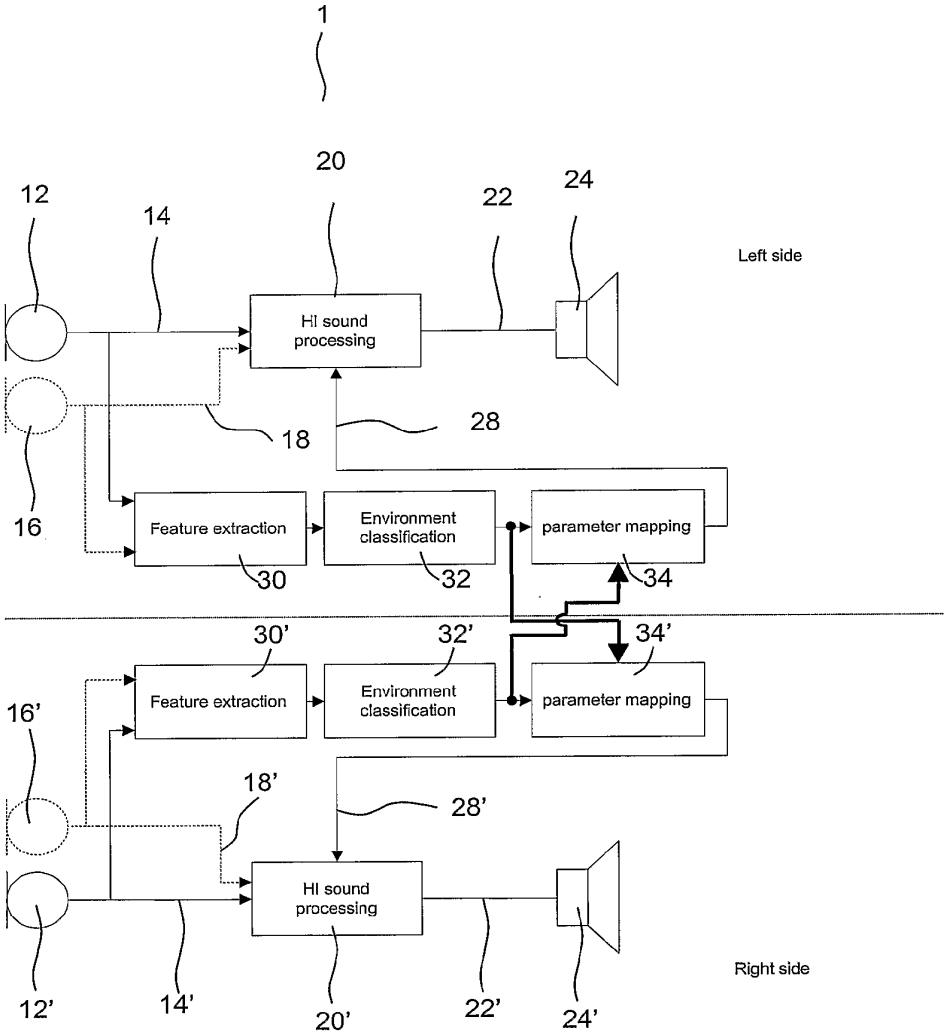

1
20
12
14
22
24
Left side
HI sound processing
18
28
16
Feature extraction
Environment classification
parameter mapping
30
32
34
30'
32'
34'
16'
Feature extraction
Environment classification
parameter mapping
18'
28'
HI sound processing
12'
14'
20'
22'
24'
Right side

Fig. 4

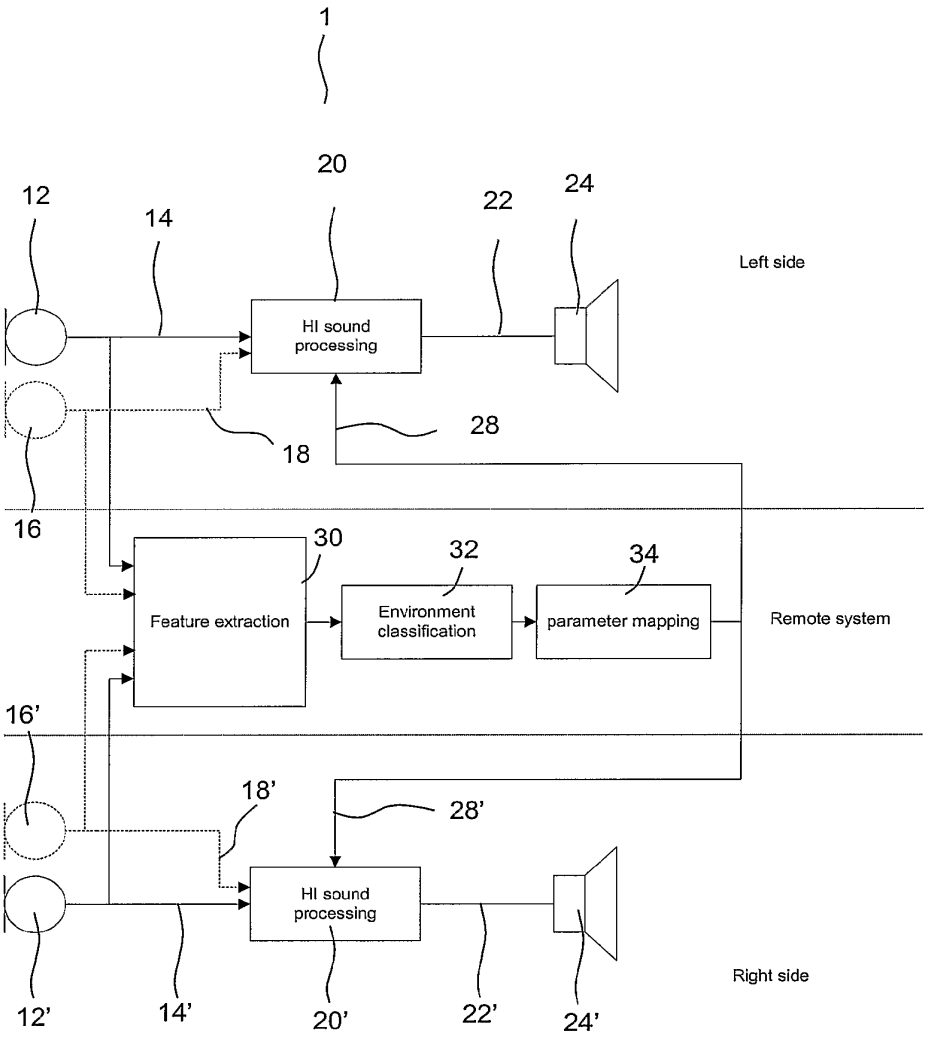

1
20
12
14
22
24
Left side
HI sound processing
28
18
16
30
32
34
Feature extraction
Environment classification
parameter mapping
Remote system
16'
18'
28'
HI sound processing
Right side
12'
14'
20'
22'
24'

Fig. 5

INTERNATIONAL SEARCH REPORT

International Application No
PCT/DK2004/000442

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04R25/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)
EPO-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

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A	US 6 549 633 B1 (WESTERMANN SOEREN ERIK) 15 April 2003 (2003-04-15) the whole document	1-8
A	US 2002/037087 A1 (ALLEGRO SYLVIA ET AL) 28 March 2002 (2002-03-28) the whole document	5
A	WO 02/32208 A (PHONAK AG ; ALLEGRO SILVIA (CH)) 25 April 2002 (2002-04-25)	

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

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

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Date of the actual completion of the international search	Date of mailing of the international search report
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Name and mailing address of the ISA	Authorized officer
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Information on patent family members

International Application No
PCT/DK2004/000442

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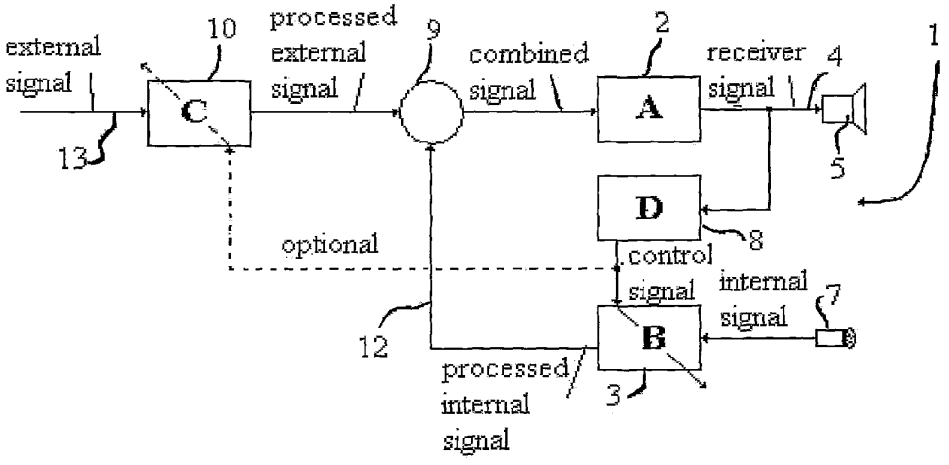
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For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) Title: ACOUSTICALLY TRANSPARENT OCCLUSION REDUCTION SYSTEM AND METHOD



(57) Abstract: A system and method that reduces the perceptual effect resulting from ear occlusion is disclosed. It comprises of an electro-acoustic feedback network that produces phase cancelling sounds in the ear, where a receiver and a microphone are located. A novel control mechanism is disclosed that controls the response of the feedback network to minimise distortion in the ear while maintaining a desired frequency response for external signals. Devices producing the external signal include hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protectors. The integration of the present invention with these devices improves the user's perception of their own voice.

WO 2006/037156 A1

Acoustically Transparent Occlusion Reduction System and Method

Field of Invention

The present invention relates to an improved occlusion reduction system for applications such as hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protection devices.

Description of Background Art

An electro-acoustic negative feedback scheme was originally presented by H.F Olson, in 1961 in U.S Patent No 2,983,790. A more comprehensive implementation was later proposed by Bose et al, in 1982 in U.S Patent No 4,494,074 under the title "Feedback Control". Using headphones, Bose proposed the idea of an electro-acoustic feedback in the proximity of the ear canal. The concept was later used by Langberg et al, in 1988 in U.S. Patent No 4,985,925, describing a system functioning as a bilateral transducer drive with a shunt feedback correction network. Later on in 1991, Langberg et al, in U.S Patent No. 5,267,321 entitled "Active Sound Absorption" describes an electro-acoustical feedback system, with the receiver acting as both a diaphragm actuator and motion sensor. In 1996, U.S Patent No. 5,774,565 to Benning et al describes an electro-acoustic feedback subsystem with oscillation prevention circuit in the forward path of the loop. In 2002, U.S Patent Application No. 2003/0012391 A1 to Armstrong et al, entitled "Digital Hearing Aids System" discloses a hearing aid including an occlusion processing subsystem.

The occlusion effect is commonly described as a hollow or echoing like sound of a person's own voice. The occlusion effect results from acoustically sealing or partially sealing the ear, or to a greater effect sealing or partially sealing the ear canal from the external acoustic environment. As a result, the occlusion effect creates discomfort and/or an unnatural sound sensation. This problem is commonly reported to clinicians in the

hearing aid industry as it affects a large number of hearing aid wearers (those with mild low-frequency hearing loss). Until now, there were at least two common schemes to decrease the occlusion effect in hearing aids, either using a vent or by increasing the insertion depth of the earmold into the ear canal. To restore naturalness of a hearing aid wearer's voice, vents of up to 3.5 mm in diameter may be employed. These vents need to be sufficiently large so that the residual sound pressure in the canal due to the occlusion effect is not significant. On the other hand, a sufficiently large vent limits the hearing aid amplification due to oscillations created by positive feedback occurring around the loop defined by an external microphone, amplifier, receiver, and path through the vent back to the external microphone.

The soft tissue in the ear canal is excited by vibration energy propagated by the skull and jaw due to the wearer's voice and this results in an increased sound pressure within the occluded of partially occluded ear relative to an open ear. Another solution is to insert the earmold further into the ear canal to fill the cartilaginous portion of the canal and hence reduce the occlusion effect at its source. However there are a number of practical problems relating to the deep insertion of an earmold, for instance physical discomfort.

While some of the aforementioned patents include methods to alter the response of the electro-acoustic feedback system, none of these patents incorporate methods to compensate for the effect that response alteration within the feedback system has on external signals such as from an external microphone or hearing aid processor. In addition, the aforementioned patents do not provide a mechanism to prevent the receiver from overloading by sensing the level of signal at the optimum point, being the input to the receiver.

Summary of Invention

It is an object of the present invention to provide an improved occlusion reduction system for applications such as hearing aids, personal sound devices, in ear monitors, communications headsets and hearing protection devices.

In accordance with a first aspect of the present invention, there is provided an electro-acoustic system comprising of: an electro-acoustic circuit including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit, (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A), for modifying the combined signal to produce an receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal such as from an external microphone or hearing aid processor to produce an external processed signal; a fourth electronic circuit (D), for estimating the level of the receiver signal optimally from the receiver signal and producing a control signal; wherein the processing performed by the second electronic circuit (B) preferably is controlled by the control signal.

The negative-feedback loop preferably can include a filter that alters the open loop response so that the real component of the response is large and negative at frequencies where the occlusion effect can be typically the greatest. Consequently reducing all signals introduced into the closed loop within this frequency range.

The third electronic circuit (C) preferably can include filtering that provides compensation for the closed loop response of the negative-feedback loop.

Optionally the filtering provided by the third electronic circuit (C) adapts to compensate for changes in the closed response of the negative-feedback loop.

Optionally the control signal can control the response of the first electronic circuit (A) while controlling the response of the third electronic circuit (C).

The microphone for generating the internal signal optionally can be coupled to the occluded or partially occluded ear by a tube. The receiver optionally can be coupled to the occluded or partially occluded ear by a tube. The microphone and receiver optionally can be combined in a single unit or jointly coupled to the occluded or partially occluded ear by a common tube. At least one of the electronic circuits can be implemented digitally. At least one of the digital electronic circuits can performed signal processing at a sampling rate at least four times that of the bandwidth of the signal.

In accordance with a further aspect of the present invention, there is provided an electro-acoustic system including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A) for modifying the combined signal to produce a receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal such as from external microphone to produce a processed external signal; a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate; wherein the control signal controls the processing performed by the second

electronic circuit (B) and controls the processing performed by the third electronic circuit (C).

In accordance with a further aspect of the present invention, there is provided an electro-acoustic system including a negative-feedback loop comprising of: a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear; a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal; a combiner for combining the processed internal signal with a processed external signal to produce a combined signal; a first electronic circuit (A) for modifying the combined signal to produce a receiver signal; and a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; a third electronic circuit (C) for modifying an external signal to produce a processed external signal; a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate; wherein the control signal controls the processing performed by first electronic circuit (A) and controls the processing performed by the third electronic circuit (C).

In accordance with a further aspect of the present invention, there is provided a method of providing a negative feedback loop for an electro-acoustic system, the method including the steps of: (a) generating an internal signal representing an acoustical signal located within or closely coupled to an occluded or partially occluded ear; (b) modifying the internal signal to produce a processed internal signal; (c) combining the processed internal signal with a processed external signal to produce a combined signal; (d) modifying the combined signal to produce a receiver signal; (e) generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear; (f) modifying an external signal to produce a processed external

signal; (g) estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate; wherein the degree of modification in step (b) is controlled by the control signal. The control signal can be utilised to control the amount of modification occurring in step (f).

In any of the aforementioned aspects of the present invention the fourth electronic circuit (D) optionally produces a control signal using estimates of the signal levels from the internal signal or the processed internal signal and the receiver signal or the combined signal.

Brief Description of the Drawings

Preferred embodiments of the present invention will now be described with reference to the accompanying drawings in which:

Fig. 1 is a block diagram of the occlusion reduction scheme of the preferred embodiment;

Fig. 2 is a schematic diagram of an analog electronic circuit for the electro-acoustic negative feedback loop;

Fig. 3 is a schematic diagram of an analog adaptive pre-compensation electronic circuit;

Fig. 4 is a schematic diagram of an analog level estimator and control signal generation electronic circuit;

Fig. 5 is a schematic diagram of a digital implementation of the occlusion reduction scheme;

Fig. 6 is a schematic diagram of an adaptive IIR filter;

Fig. 7 is a schematic diagram of a level estimator and control signal generator ;

Fig. 8 is a graph of the simulated closed loop response in a Zwislocki coupler;

Fig. 9 is a graph of the measured response of the effective gain reduction inside a real ear;

Fig. 10 is a graph of the measured responses from an external signal to a Zwislocki coupler microphone; and

Fig. 11 is an illustration of the occlusion reduction scheme implemented as an in-the-canal hearing aid device.

Detail Description of the Preferred and Other Embodiments

The preferred embodiment operates to reduce the level of signals generated within an electro-acoustic negative feedback loop, such as signals produced by vibration in the ear canal walls due to bone conduction of a user's voice. The reduction occurs in the low to mid audible frequencies, typically ranging from 80 Hz up to 1 kHz, where the occlusion effect is more predominant and perceptually apparent.

A negative feedback scheme is provided which combines a processed externally generated signal such as from an external microphone or a sound system with a processed internal signal such as from a microphone located within or closely coupled to the occluded or partially occluded ear. The combined signal after optional further processing is applied to a receiver located within or closely coupled to the occluded or partially occluded ear. The level of the signal to be applied to the receiver is optimally estimated either from the signal applied to the receiver or from the combined processed external and processed internal signals. Optionally this signal level can be estimated from signals at other points within the scheme. High signal levels applied to the receiver may produce a distorted output from the receiver. This distortion is reduced by applying active gain reduction in the feedback path in response to estimated high signal levels being present. Optionally low signal levels can be detected so that noise inherent in the negative feedback components such as the internal microphone can be minimised by

applying active gain reduction in the feedback path. Thus, the high and the low signal level thresholds for gain reduction in the loop can depend on the dynamic operational range of the discrete components within the system. In addition, an adaptive equalisation filter is applied to the external signal to compensate for variations of the transfer response of the closed loop.

The preferred embodiment includes of a microphone to sense the sound pressure in the ear. The preferred embodiment also includes a novel design of estimating the level of the signal to be applied to the receiver and reducing the gain in the feedback path of the loop when this level is high. This mechanism effectively improves the robustness of the closed loop system by limiting excessive driving levels being applied to the receiver. A filter within the feedback path of the loop yields the necessary phase and gain around the loop to generate a phase cancelling sound in the ear without creating acoustic feedback. This negative feedback response also causes a sound pressure reduction for external signals thus affecting the response from the external processed signal to the receiver signal. As a result, an adaptive pre-compensation filter is provided. The adaptive pre-compensation filter performs adaptive equalisation to maintain a constant frequency response between the external signal and the receiver signal in response to changes in loop response.

In addition, the negative feedback response also causes a reduction in external sounds transmitted through a vent or leakage, thus minimising both effects.

Referring initially to **Fig. 1**, a schematic diagram 1 illustrates the occlusion reduction scheme of the preferred embodiment. This electronic circuit can be encapsulated in an earmold 100 as discussed hereinafter with reference to **Fig. 11**. The earmold optionally contains tubing for coupling the receiver to the ear. The earmold optionally contains tubing for coupling the internal microphone to the ear. The

microphone and receiver optionally can be combined into a single unit or jointly coupled to the occluded or partially occluded ear by a common tube. Optionally these tubes can be protected from wax blockage using wax guards. The lengths of these tubes are preferably as short as possible to minimise delays around the feedback loop, but can be any length. The earmold can optionally contain an open vent to depressurise the ear thus reducing the sensation of stuffiness in the ear. Optionally, the vent can be fitted with an acoustic damper for compensating for the vent resonance that may affect the closed loop response.

The internal signal from an internal microphone 7 is proportional to the ear canal sound pressure. This internal signal is filtered in a feedback loop shown as first electronic circuit **(A)** 2 and second electronic circuit (**B) 3** to produce the receiver signal 4 output to the receiver 5. The aim is to produce cancellation around the loop, limited to a given low to mid frequency band.

An analog implementation of the occlusion reduction scheme is depicted in **Figs. 2-4** and a digital implementation is shown in **Figs. 5-7**.

In addition, **Fig. 1** shows fourth electronic circuit (D) 8 that estimates the level of the receiver signal to produce a control signal. Optionally the input to the fourth electronic circuit (D) 8 can be obtained from the output of a combiner 9 with appropriate compensation for the effects of the first electronic circuit (A) 2. The response of the negative feedback loop is controlled by the control signal. The estimated level of the receive signal 4 can be compared to a reference level which is not shown in this figure. The control signal reduces the gain in the loop as the level of the receive signal increases above the reference level. The reference level is set to a level to minimise distortion occurring within the loop.

The combined signal within the loop results from a combination 9 of processed internal signal 12 and processed external signal. The processed external signal results from a filtered external signal 13. The pre-compensation filter 10 depicted as the third electronic circuit (**C**) 10 in **Fig. 1**, equalises the magnitude of the transfer function from the external signal 13 to the receiver signal 4 so that it is approximately constant across frequency, assuming a fixed closed loop response. Optionally, the filter in the third electronic circuit (C) 10 is adaptively controlled by fourth electronic circuit (D) 8 so that the magnitude of the transfer function from the external signal 13 to the receiver signal 4 is approximately constant across frequency regardless of changes in the closed loop response.

Optionally the fourth electronic circuit (D) uses estimates of signal levels from the receiver signal or combined signal and from the internal signal or processed internal signal to produce a control signal to control the loop response. This control signal can control the loop response by directly applying gain reduction to the first electronic circuit (A) or to the second electronic circuit (B). The control signal produce from this arrangement can be used in combination with the fixed or adaptive third electronic circuit (C).

Analog Implementation

Fig. 2, depicts an analog implementation 20 of the negative feedback loop of **Fig. 1**. Firstly, the internal signal is applied to a high pass filter 21 and lead-lag filter 22 in order to reduce effects from jaw movements and very low frequency instability in the loop. The equations for the transfer function in relation to the corresponding discrete components are shown below:

$$H_{HP}(j\omega) = \frac{j\omega \cdot R_1 C_2}{1 + j\omega \cdot R_1 \times C_1}$$

if $R_4 = R_5, C_2 = C_3$ then the lead lag transfer function equation is shown below

$$H_{Lead-Lag}(j\omega) = \frac{(R_3 + R_6) + j\omega \bullet C_2 \bullet (2R_3R_6 + R_6R_4 + R_3R_4)}{R_6 + j\omega \bullet C_2 \bullet (R_3R_6 + R_6R_4)}$$

The next filtering stage reduces the dominant transducer resonance within the loop and provides greater open loop gain at frequencies at which the occlusion effect is greatest. This is achieved by using a bi-quadratic filter 24, and placing a complex pair of zeros at the dominant transducer resonance frequency followed by a pair of real poles to provide low frequency emphasis in the open loop response.

In the arrangement of Fig. 2, if we allow $R_8 = R_{10}, C_4 = C_6$, $R_9 = R_{11}, C_5 = C_7$ and $R_{12} = R_{13}, C_8 = C_9$,then the biquadratic transfer function equation can be computed as follows

$$H_{bi-quad}(j\omega) = \frac{(j\omega)^2 C_4 R_9 C_5 + j\omega\left(C_4 + \frac{C_5 R_9}{R_8} - C_5\right) + \frac{1}{R_8}}{(j\omega)^2 C_5 R_9 C_8 + j\omega\left(\frac{C_5 R_9}{R_{12}} + C_8\right) + \frac{1}{R_{12}}}$$

To determine the pole zero placement, the biquadratic transfer function equation can be directly related to a well known second order transfer function equation.

$$K(j\omega) = \frac{(j\omega)^2 + 2\xi_z \omega_z j\omega + \omega_z^2}{(j\omega)^2 + 2\zeta_p \omega_p j\omega + \omega_p^2}$$

Where the ω_p and ω_z are the location of the pole and zero frequencies. Similarly ξ_z and ζ_p are the damping factors for the poles and zeros respectively. Thus, this relationship can be used to position the poles and zeros at the desired frequencies with the desired damping.

With appropriate phase compensation, amplification is added to the loop. The gain amplification is chosen in accordance to a well-known gain and phase margin criteria (e.g. Linear Control Systems Analysis and Designs, John J.D'Azzo, Constantine H.

Houpis, 2nd Edition, McGraw-Hill, 1981). A loop gain of less than or equal to –3 dB is chosen at frequencies likely to produce positive feedback. The filtering arrangement of **Fig. 1** shown in **Fig 2**, produces a gain greater than unity, for an open loop response between 80 Hz and 1 kHz and less than unity at other frequencies, where positive feedback may occur. The frequency band ranging from 80 Hz up to 1 kHz is an appropriated choice as the occlusion effect is subjectively more apparent at these frequencies, as described in literature (e.g. Hearing Aids, Harvey Dillon, Boomerang-Press, 2001).

The bi-quadratic filter 24 is followed up by a voltage controlled variable resistance 25, referred to as **VVR**. This control produces up to – 20 dB of gain around the loop, by controlling the voltage at the gate of the JFET. Thus, the gain across the VVR network is found by observing that the JFET is series with R_7 forms a voltage divider, with its gain given by:

$$Gain_{VVR} = \frac{R_{Q1}}{R_7 + R_{Q1}}$$

Where R_{Q1}, is the variable resistance across drain to source junction.

The VVR is followed by a bufferstage 26, that is subsequently followed by a variable resistor, R_{14}. 27. This latter resistor is used to fined tune the gain around the loop manually. The variable resistor 27 is followed by an amplifier and a combiner 9, combining a processed external signal 28 with the processed internal signal 12. Finally, this buffer functions as a voltage controlled voltage source to the receiver, optionally a **class D** amplifier may be used.

Referring now to **Fig. 3**, there is illustrated the precompensation circuit 10 of **Fig. 1** in more detail. The circuit may be used to pre-compensate an external signal 13 to produce a processed external signal 28. The external signal is pre-compensated with a

fixed high pass filter 32, a variable notch filter 33, and an adjustable gain control 34. Note that the notch and the gain are also controlled with voltage variable resistances, using JFET transistors.

If $C_{13} = C_{14}$,then the transfer function equation for the notch filter, positioned between U6 and U7 is shown below

$$H_{Notch}(j\omega) = \frac{1 + j\omega(2R_x C_{13}) + \left(R_x R_{19} C_{13}^{\ 2}\right)\cdot(j\omega)^2}{1 + j\omega(R_{19} + 2R_x)C_{13} + \left(R_x R_{19} C_{13}^{\ 2}\right)\cdot(j\omega)^2}$$

Where $R_x = R_{20} + R_{Q3} + R_{Q4}$ and $R_{Q3} + R_{Q4}$ is the combined resistance across drain to source junction of the two JFET transistors. Also note that the amplification at U7 is controlled by the source to drain resistance at Q5, and the low pass filter between U5 and U6 is essentially switched on and off by Q2.

The voltage variable resistances shown in **Fig. 2** and **3** are driven by the circuit 40 shown in **Fig. 4**. This circuit functions as a signal level estimator and threshold detector and produces a control voltage 41. In this circuit the receiver signal 4 is compared to a reference voltage 43, shown as v-Ref in the figure, at the negative input of U8 (44). If this voltage does not exceed the reference level, the current source, made up by BJ3 and BJ4 transistor pair pulls down the voltage across the RC tank towards the negative supply voltage, with a time constant equal to the release time. Note that the voltage across the RC tank feeds directly into the gate of the JFET control transistor in **Fig. 2** from U9 buffer (1), resulting in a VVR signal 25. Similarly from U10 and U11 buffer (2) and buffer (3), results is an inverted VVR signal 35, that is directly applied to the gates of all the JFET transistors shown in **Fig. 3**. As the voltage exceeds the threshold reference level, the current source, shown as BJ6 and BJ5 transistor pair, charges the RC tank, towards the base voltage at BJ6, with a time constant set by the attack time. The

attack and release time constants can be approximated by finding the appropriate ratio between resistors R_{31} and R_{30}.

The voltage across the capacitor C_{15} ranges from a voltage at the based of BJ7 , say 0.7V or fully charged to the voltage at the collector of BJ3, say -2.8V or fully discharged, then the change in voltage $\Delta V \approx 3.5V$, from which the current needed to produce the discharging rate can be evaluated as follows:

$$\text{Discharging } I = C_{15} \times \frac{\Delta V}{\text{Re}lease\_Time} = 7uA$$

thus effectively $I_{charging} \approx 11 \times I_{discharging}$

Then $R_{31} = \frac{0.7}{77u} \approx 8.2k$ $\qquad R_{30} = 11 \times R_{31} \approx 91k$

Digital Implementation

A digital implementation of the arrangement of **Fig. 1** typically requires the signal within the loop to be oversampled to reduce the delay introduced into the loop by sampling. The signal processing performed by electronic circuits A, B, C and D and the combiner 9 may be performed digitally. The digital processing optionally is performed by a digital signal processor which is instructed to perform the signal processing. Various techniques are available to those skilled in the art to perform these processes. Examples of these are performing the filtering operations using FIR or IIR filters or modifying the signals in the frequency domain using the short-time Fourier transform techniques. One implementation of the scheme described for **Fig 1** is shown in **Figs. 5-7**.

In **Fig. 5,** after A/D conversion 50, the internal signal is filtered with an IIR filter 51 in order to provide the required gain and phase compensation, with the same criteria applied in the description of **Fig. 2**.

In this digital implementation the level of the receiver signal 4 is directly estimated and the control signal *r(n)* is produced by the level estimator and control signal generator

8 which is shown in greater detailed in **Fig. 7.**. Referring now to **Fig. 7,** the receive signal level is estimated with a digital 1st order switchable time-constant envelope detector. The envelope signal is compared 71 to a threshold value and the maximum of either the ratio of its exceedance of this threshold or unity is produced. This maximum is inverted to produce the control signal. As the level of the envelope signal exceeds the threshold, the control signal shown as *r(n)* decreases at a rate determined by the attack time constant λ_{attack}. As the envelope signal decreases in level, the control signal increases at a rate determined by the release time constant $\lambda_{release}$. Alternative signal level compression strategies employing different envelope detector designs well known to those skilled in the art may be used.

The internal signal, after being filtered by the IIR filter 51, is scaled by the control signal *r(n)* to produce the processed internal signal. The processed external signal is added to the processed internal signal by the combiner 9 to produce the combined signal. The combined signal is scaled 2 to produce the receiver signal 4 which after D/A conversion is applied to the receiver 5.

As in the analog description, the negative feedback loop requires pre-compensation so that the gain response measured from the external signal to the receiver signal is not altered by the closed loop response. In **Fig. 5,** the pre-compensation filter 10 is realised as a cascaded IIR filter, with a structural realisation shown in **Fig. 6** . The first filter stage shown in **Fig. 6** is a high pass filter, comprising of a zero at 0 Hz and a pole at f_{HP}, the coefficient being a function of the sampling rate, Fs. The second stage of the filter shown in **Fig. 6** is adaptive, the locations of poles f_p and zeros f_z of the digital filter are varied in accordance with the output of a mapping function *p(n)* 61. Alternatively the damping factors for the poles and zeros can be controlled by two independent mapping functions derived from the control signal *r(n)*. **Fig. 5** shows *p(n)*

as a single output from a polynomial mapping function with input *r(n)*. The polynomial's coefficients are found using an autoregressive analysis of the required compensating response as a function of the control signal *r(n)*.

Any combination of analog and digital electronic circuits is possible in addition to the all analogue and the all digital implementations described with the appropriate conversions between analogue and digital formats. In particular, electronic circuits B, C and D may be implemented digitally with the combiner and electronic circuit A implemented in analog circuitry. Furthermore, the processed internal and processed external signals may be combined while both in a 1-bit format and applied directly to the receiver without electronic circuit A performing any function or with it simply providing the 1-bit drive current for the receiver.

Referring now to **Fig. 8** the simulated closed loop response in a Zwislocki coupler is shown. The scheme yields approximately 16 dB of occlusion reduction at 300 Hz, and 18 dB at 100 Hz. The occlusion reduction response is a measure of the level difference between the sound level in the coupler with the occlusion reduction system active and the sound level in the coupler without any occlusion reduction. Referring to **Fig. 9** occlusion reduction in a real ear is shown. This was calculated by taking the linear average of the energy in 1/12 octave bands recorded using an in-the-ear-canal microphone while a subject talked for 2 minutes. This signal was recorded while the feedback loop gain is maximum, and then again while the loop was open.

Referring now to **Fig. 10**, three responses measured in a Zwislocki coupler are shown. These are the transfer functions from an external signal to the coupler microphone for the compensated 93 and un-compensated responses 92. Also shown is the natural receiver response 94 employing a voltage controlled voltage source to the receiver. Note that the compensated response is not adversely affected by the occlusion

reduction scheme compared to that of the natural receiver response 94. Also similar responses are derived for minimally open vented cavities with the appropriate changes to the mappings, filtering and gain equalisation in electronic circuits A, B, and C.

Referring now to **Fig. 11**, there is shown one form of actual implementation of the present invention as an in-the-canal hearing aid device 100 placed in a user's ear 101 that substantially eliminates the discomfort associated with the occlusion effect. The device 100 includes a sealed or optionally vented 102 earmold that occludes or partially occludes the ear canal. The earmold encapsulates a hearing aid electronic circuit comprising of: a hearing aid processor 103 directly connected to an external microphone 104 and powered by small battery 105. The hearing aid processor includes occlusion reduction electronic circuits A, B, C, D and the combiner. A separate receiver 5 and an internal microphone 7 are also interconnected. These transducer elements are acoustically coupled to the inside of the ear canal with short tubes 108, 109; and protected from ear wax with wax-guarding devices 110, 111.

The forgoing describes preferred embodiments of the present invention. Modifications, obvious to those skilled in the art, can be made thereto without departing from the scope of the invention.

CLAIMS

1. An electro-acoustic system including:

an electro-acoustic circuit including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate;

wherein the processing performed by the second electronic circuit (B) is controlled by the control signal.

2. A system as claimed in claim 1 wherein the negative-feedback loop includes filtering that alters the open loop response so that the real component of the response is large and negative at frequencies where the occlusion effect is most apparent.

3. A system as claimed in claim 2 wherein the third electronic circuit (C) includes filtering that provides compensation for the closed response of the negative-feedback loop.

4. A system as claimed in claim 3 wherein the filtering provided by the third electronic circuit (C) is adaptive and controlled by the control signal.

5. A system as claimed in any previous claim wherein the fourth electronic circuit (D) estimates the level of the internal signal or the level of the processed internal signal and uses this estimate with the estimate of the receiver signal to produce a control signal.

6. A system as claimed in any previous claim wherein the gain of the second electronic circuit (B) is controlled by the control signal.

7. A system as claimed in any previous claim wherein the ear is occluded or partially occluded by an earmold.

8. A system as claimed in claim 7 wherein the earmold contains a vent.

9. A system as claimed in claim 8 wherein the vent contains an acoustic damper

10. A system as claimed in any previous claim wherein the microphone for generating the internal signal is coupled to the occluded or partially occluded ear by a tube.

11. A system as claimed in any previous claim wherein the receiver is coupled to the occluded or partially occluded ear by a tube.

12. A system as claimed in any previous claim wherein the microphone and the receiver are jointly coupled to the occluded or partially occluded ear by a common tube.

13. A system as claimed in any previous claim wherein at least one of the electronic circuits is digital.

14. A system as claimed in any previous claim wherein at least one of the digital electronic circuits uses a sampling rate at least four times that of the bandwidth of the signal.

15. A system as claimed in claims 13 or claim 14 wherein the first electronic circuit (A) includes a digital to analogue converter.

16. A system as claimed in claim 13, claim 14 or claim 15 wherein the second electronic circuit (B) includes an analogue to digital converter.

17. A system as claimed in claim 16 wherein the second electronic circuit (B) includes a digital to analogue converter.

18. A system as claimed in claim 14 wherein at least one of the input signals to the combiner is a binary signal.

19. An electro-acoustic system including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal such as from external microphone to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate;

wherein the control signal controls the processing performed by the second electronic circuit (B) and controls the processing performed by the third electronic circuit (C).

20. An electro-acoustic system including a negative-feedback loop comprising of:

a microphone for generating an internal signal from an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

a second electronic circuit (B) for modifying the internal signal to produce a processed internal signal;

a combiner for combining the processed internal signal with a processed external signal to produce a combined signal;

a first electronic circuit (A) for modifying the combined signal to produce a receiver signal;

and

a receiver for generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

a third electronic circuit (C) for modifying an external signal to produce a processed external signal;

a fourth electronic circuit (D) for estimating the level of the receiver signal and producing a control signal from this estimate;

wherein the control signal controls the processing performed by first electronic circuit (A) and controls the processing performed by the third electronic circuit (C).

21. A method of providing a negative feedback loop for an electro-acoustic system, the method including the steps of:

(a) generating an internal signal representing an acoustical signal located within or closely coupled to an occluded or partially occluded ear;

(b) modifying the internal signal to produce a processed internal signal;

(c) combining the processed internal signal with a processed external signal to produce a combined signal;

(d) modifying the combined signal to produce a receiver signal;

(e) generating an acoustical signal from the receiver signal at a location within or closely coupled to the occluded or partially occluded ear;

(f) modifying an external signal to produce a processed external signal;

(g) estimating the level of the receiver signal either directly from the receiver signal or from the combined signal and producing a control signal from this estimate;

wherein the degree of modification in step (b) is controlled by the control signal.

22. A method as claimed in claim 21 wherein the control signal is utilised to control the amount of modification occurring in step (f).

23. A method substantially as herein described with reference to any one of the embodiments of the invention illustrated in the accompanying drawings and/or examples.

24. A system substantially as herein described with reference to any one of the embodiments of the invention illustrated in the accompanying drawings and/or examples.

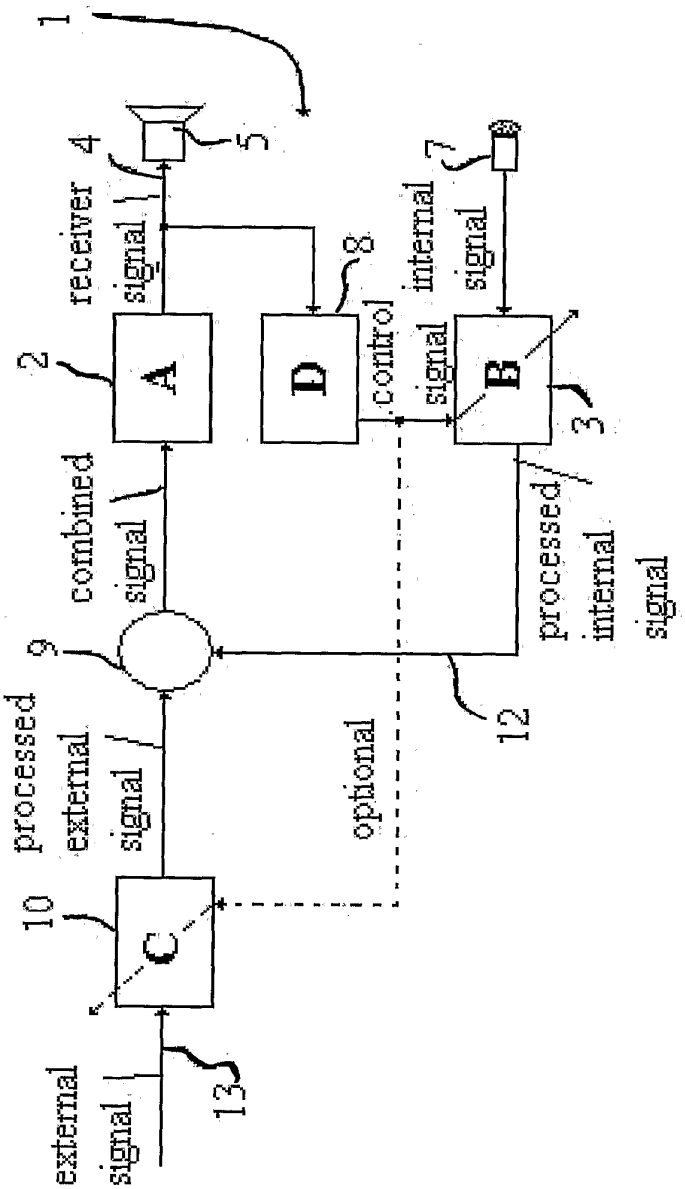

1
receiver
4
signal
5
2
A
D
8
control
signal
internal
signal
7
B
3
combined
signal
9
processed
external
signal
10
C
optional
12
processed
internal
signal
external
signal
13

Fig. 1

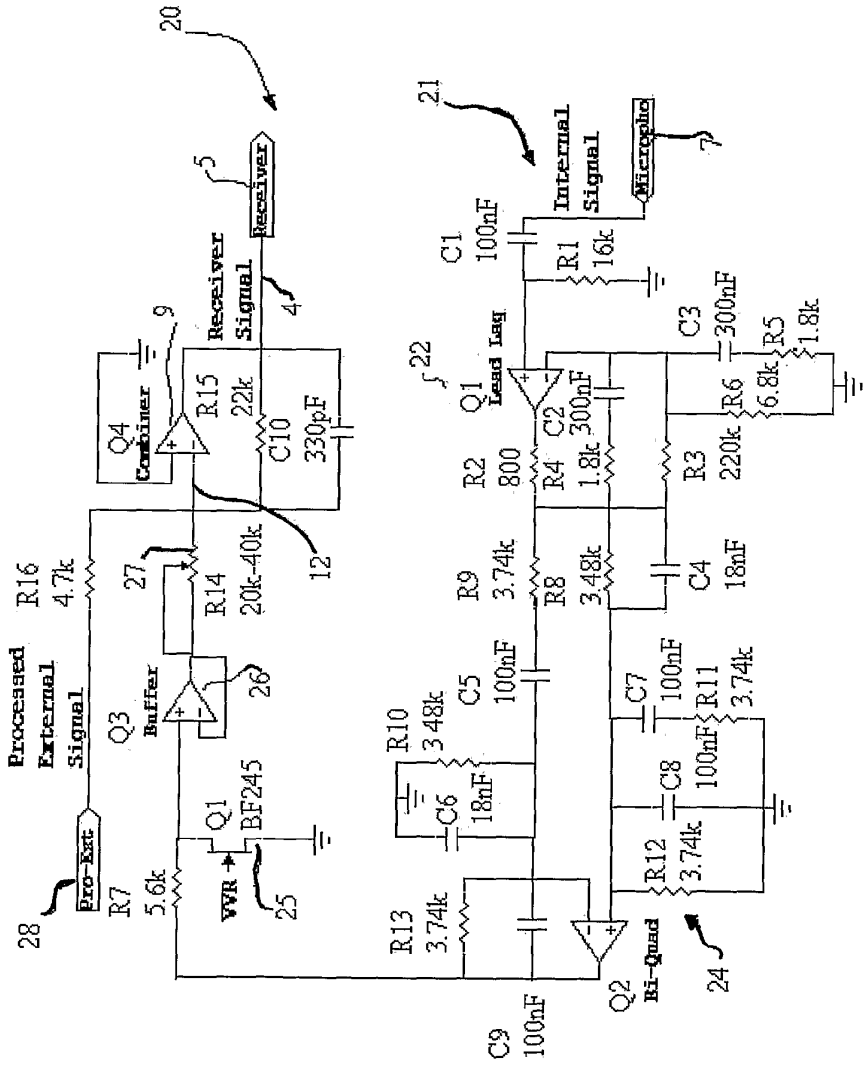

20
21
22
Receiver
Receiver Signal
Combiner
Q4
R15 22k
C10 330pF
Internal Signal
Microphone
C1 100nF
R1 16k
Lead Lag
Q1
C2 300nF
R4 1.8k
R2 800
C3 300nF
R5 1.8k
R6 6.8k
R3 220k
Processed External Signal
Pro-Ext
R16 4.7k
R14 20k-40k
Buffer
Q3
Q1 BF245
VVR
R7 5.6k
R9 3.74k
R8 3.48k
C4 18nF
R10 3.48k
C6 18nF
C5 100nF
C7 100nF
R11 3.74k
C8 100nF
R12 3.74k
R13 3.74k
C9 100nF
Bi-Quad
Q2
24
25
26
27
28
12
4
5
7
9

Fig. 2

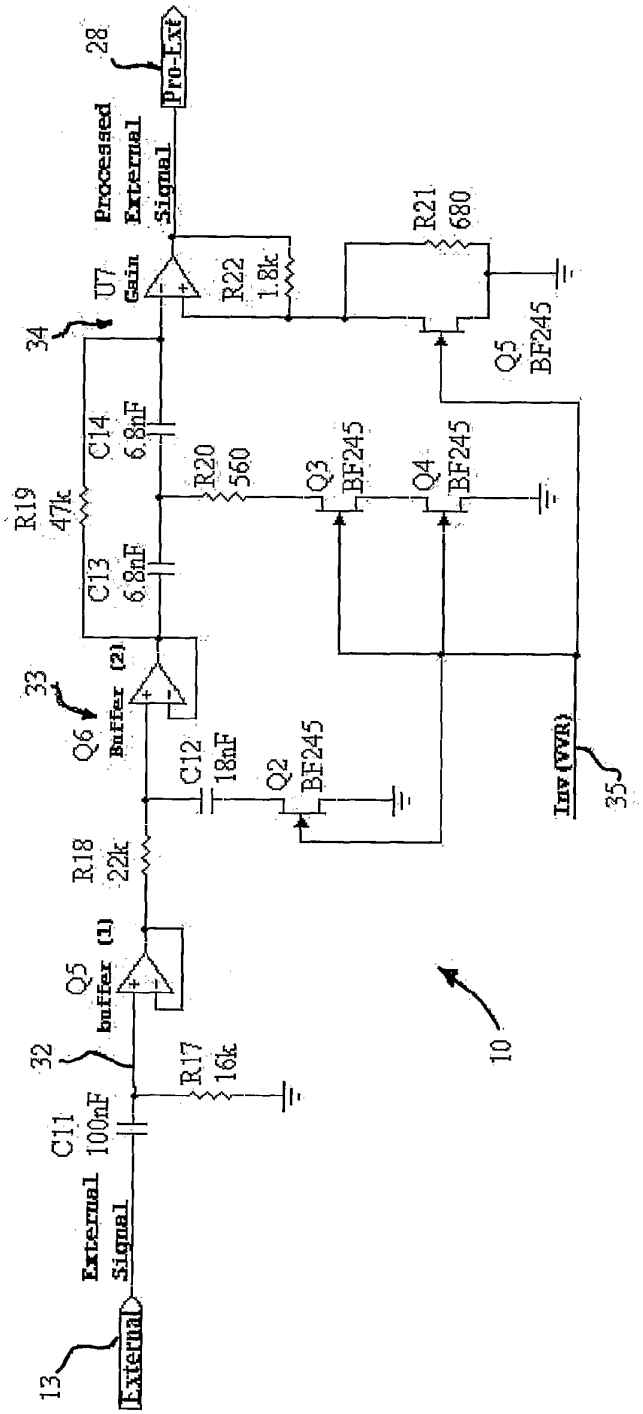

28
Pro-Ext
Processed External Signal
U7
Gain
34
R22
1.8k
R21
680
Q5
BF245
C14
6.8nF
R20
560
Q3
BF245
Q4
BF245
R19
47k
C13
6.8nF
33
Q6
Buffer (2)
C12
18nF
Q2
BF245
Inv (VVR)
35
R18
22k
Q5
Buffer (1)
32
R17
16k
C11
100nF
External Signal
External
13
10

Fig. 3

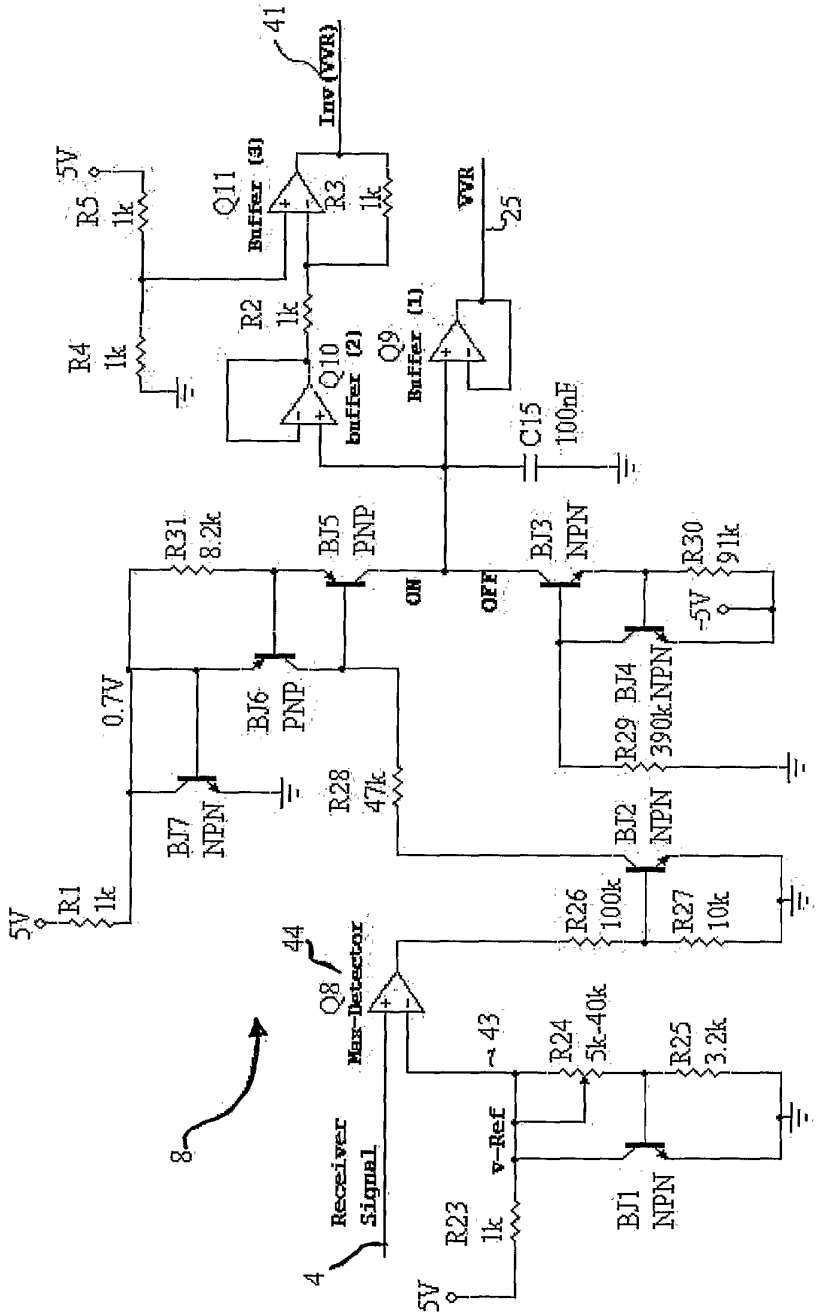

41
Inv (VVR)
5V
R5 1k
Q11 Buffer (3)
R3 1k
VVR
25
R4 1k
R2 1k
Q10 buffer (2)
Q9 Buffer (1)
C15 100nF
R31 8.2k
BJ5 PNP
ON
OFF
BJ3 NPN
R30 91k
-5V
0.7V
BJ6 PNP
BJ4 NPN
R29 390k
BJ7 NPN
R28 47k
BJ2 NPN
5V
R1 1k
R26 100k
R27 10k
44
Q8 Max-Detector
~ 43
R24 5k-40k
R25 3.2k
8
v-Ref
Receiver Signal
4
R23 1k
BJ1 NPN
5V

Fig. 4

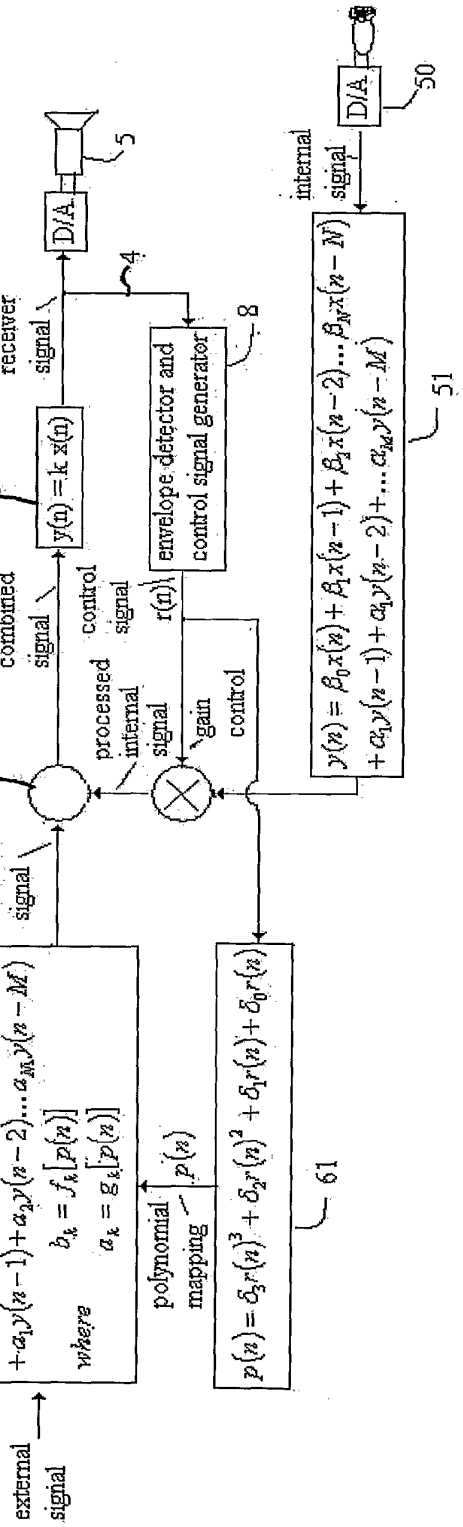

external signal
$y(n) = b_0x(n) + b_1x(n-1) \dots b_Nx(n-N)$
$+ a_1y(n-1) + a_2y(n-2) \dots a_My(n-M)$
where $b_k = f_k[p(n)]$ $a_k = g_k[p(n)]$
10
processed external signal
9
combined signal
processed internal signal
control signal r(n)
gain control
polynomial mapping
p(n)
$p(n) = \delta_3 r(n)^3 + \delta_2 r(n)^2 + \delta_1 r(n) + \delta_0 r(n)$
61
y(n) = k x(n)
2
receiver signal
D/A
5
4
envelope detector and control signal generator
8
$y(n) = \beta_0 x(n) + \beta_1 x(n-1) + \beta_2 x(n-2) \dots \beta_N x(n-N)$
$+ \alpha_1 y(n-1) + \alpha_2 y(n-2) + \dots \alpha_M y(n-M)$
51
internal signal
D/A
50

Fig. 5

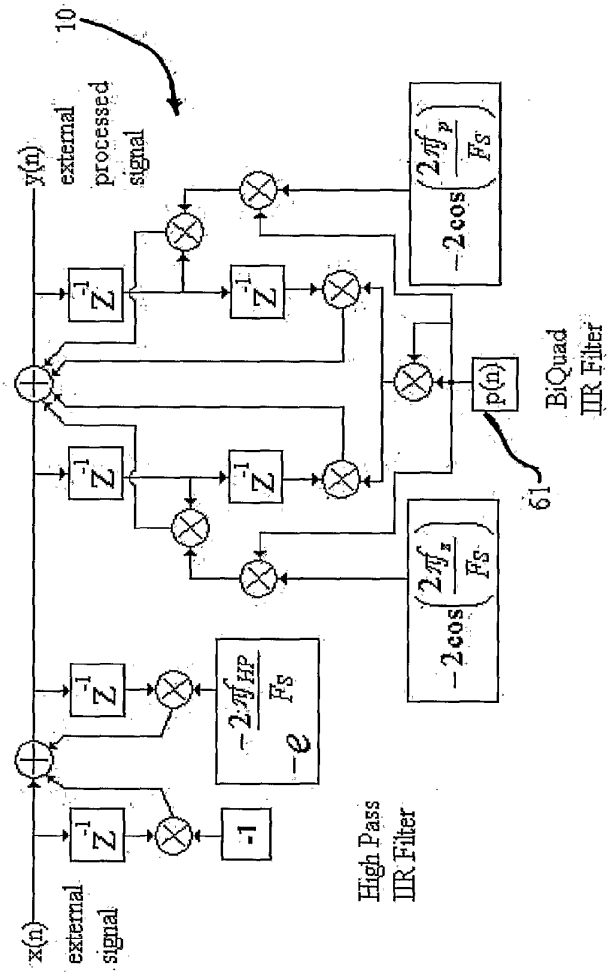

10
y(n)
external
processed
signal
x(n)
external
signal
High Pass
IIR Filter
BiQuad
IIR Filter
p(n)
61
-1
$-2\cos\left(\frac{2\pi f_p}{F_S}\right)$
$-2\cos\left(\frac{2\pi f_z}{F_S}\right)$
$-e^{-\frac{2\pi f_{HP}}{F_S}}$

Fig. 6

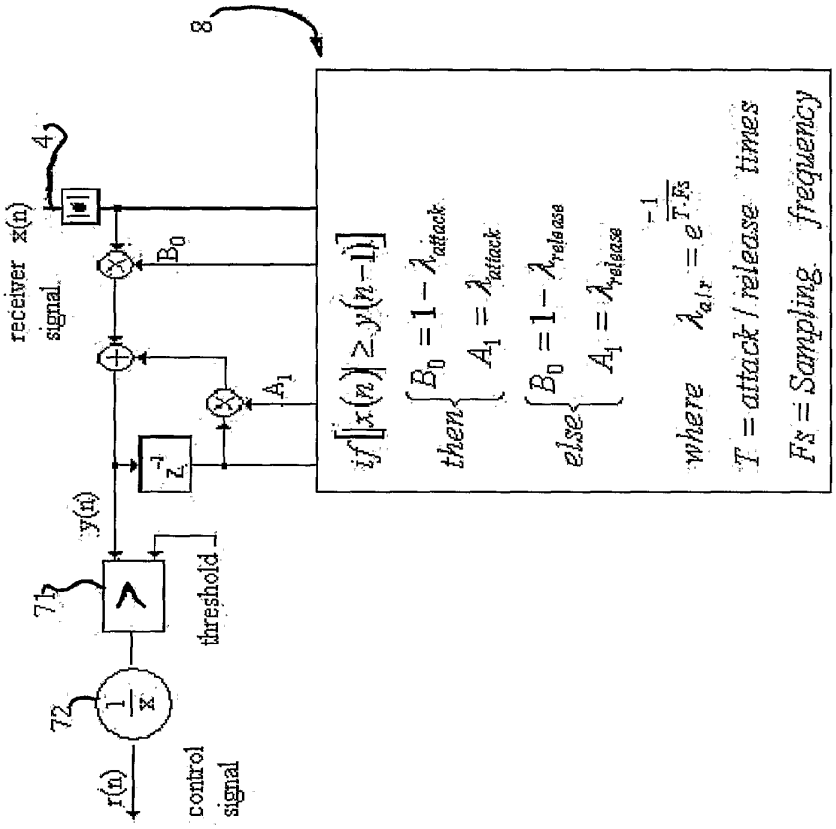

8
4
receiver x(n)
signal
|x|
B_0
A_1
z^{-1}
y(n)
7b
>
threshold
72
$\frac{1}{x}$
r(n)
control
signal
$if\ [|x(n)| \geq y(n-1)]$
$then \begin{cases} B_0 = 1 - \lambda_{attack} \\ A_1 = \lambda_{attack} \end{cases}$
$else \begin{cases} B_0 = 1 - \lambda_{release} \\ A_1 = \lambda_{release} \end{cases}$
$where \quad \lambda_{a/r} = e^{\frac{-1}{T \cdot Fs}}$
$T = attack\ /\ release\ \ times$
$Fs = Sampling\ \ frequency$

Fig. 7

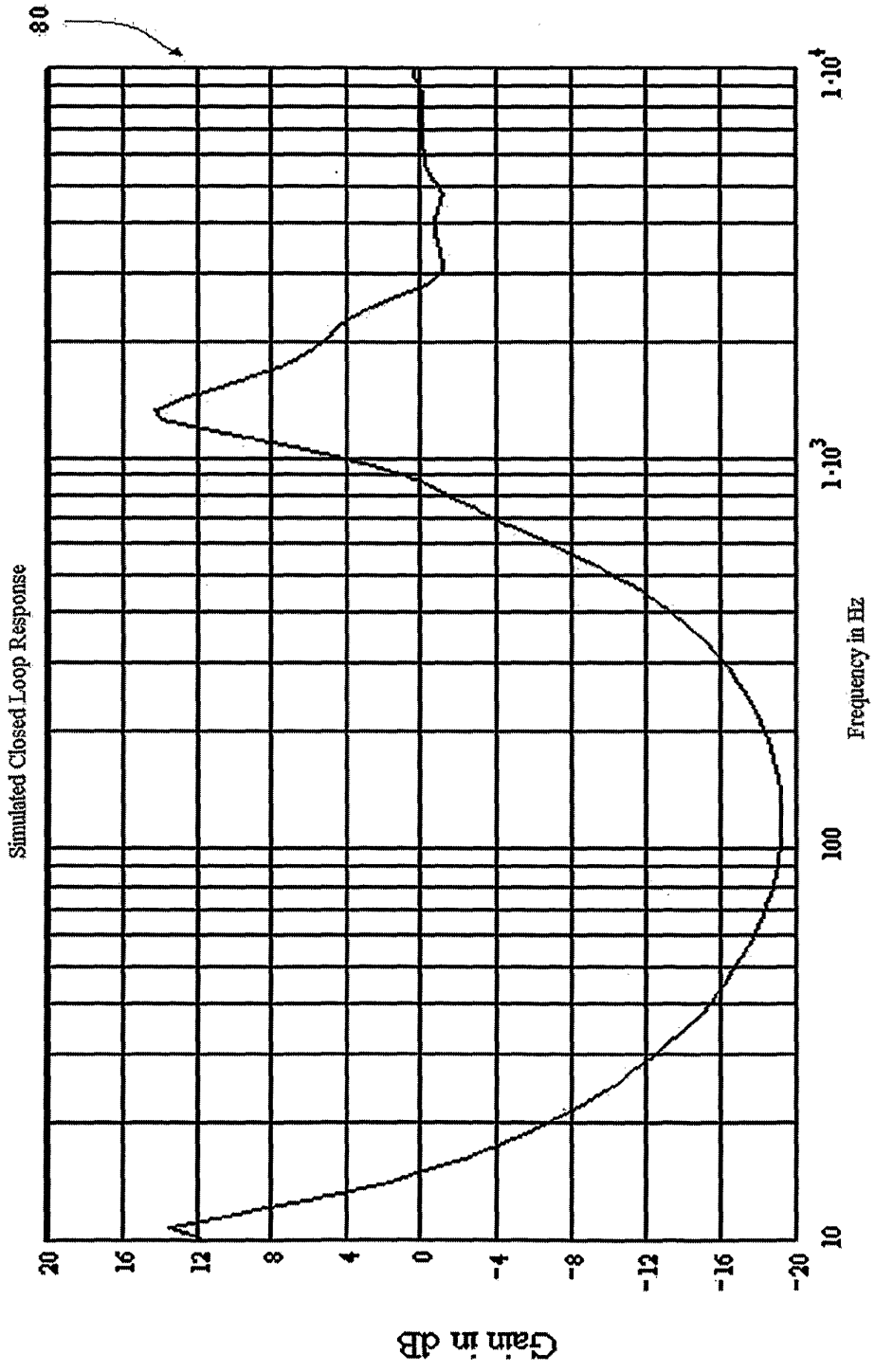

80
Simulated Closed Loop Response
Gain in dB
20
16
12
8
4
0
-4
-8
-12
-16
-20
10
100
1·10^3
1·10^4
Frequency in Hz

Fig. 8

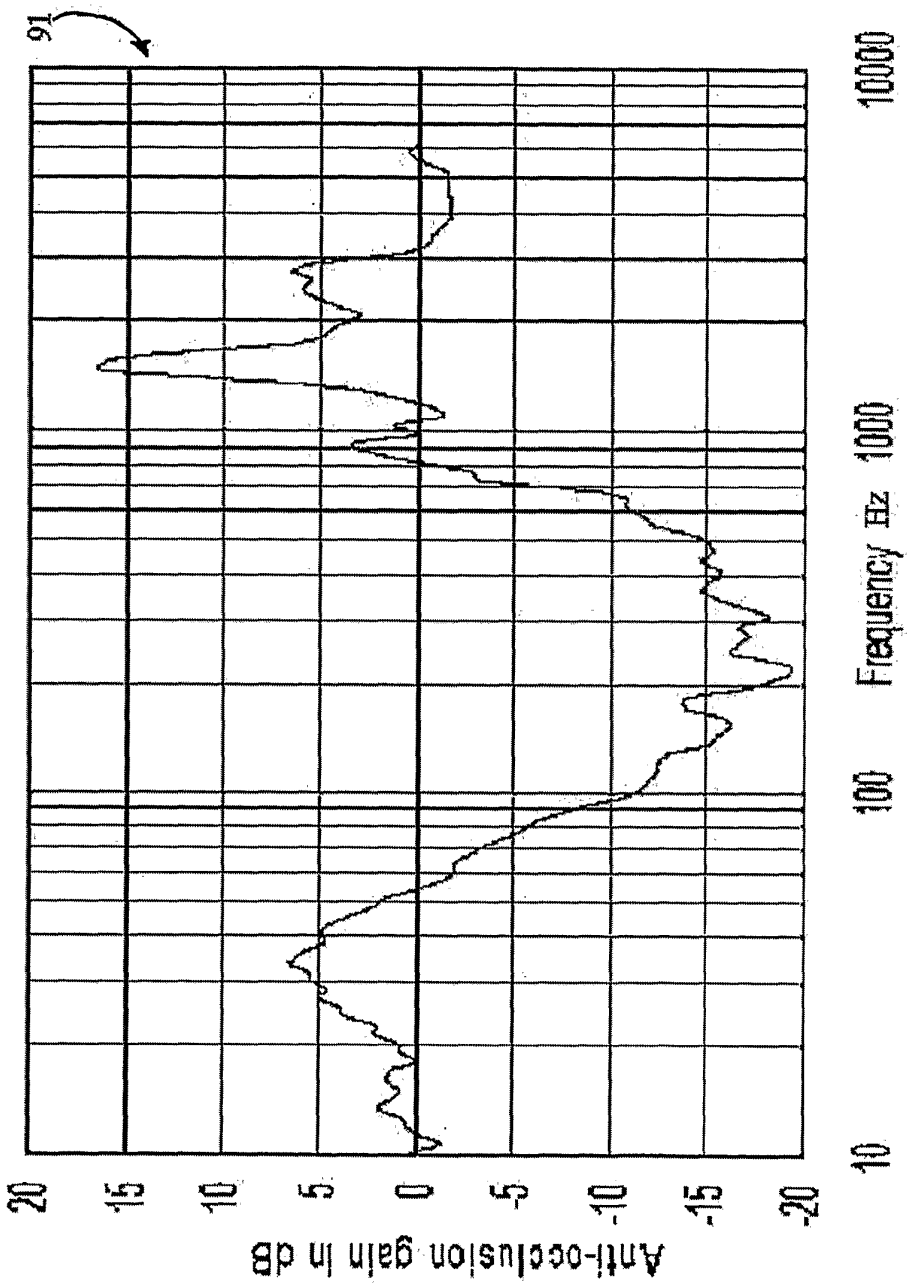

91
Anti-occlusion gain in dB
20
15
10
5
0
-5
-10
-15
-20
10
100
1000
10000
Frequency Hz

Fig. 9

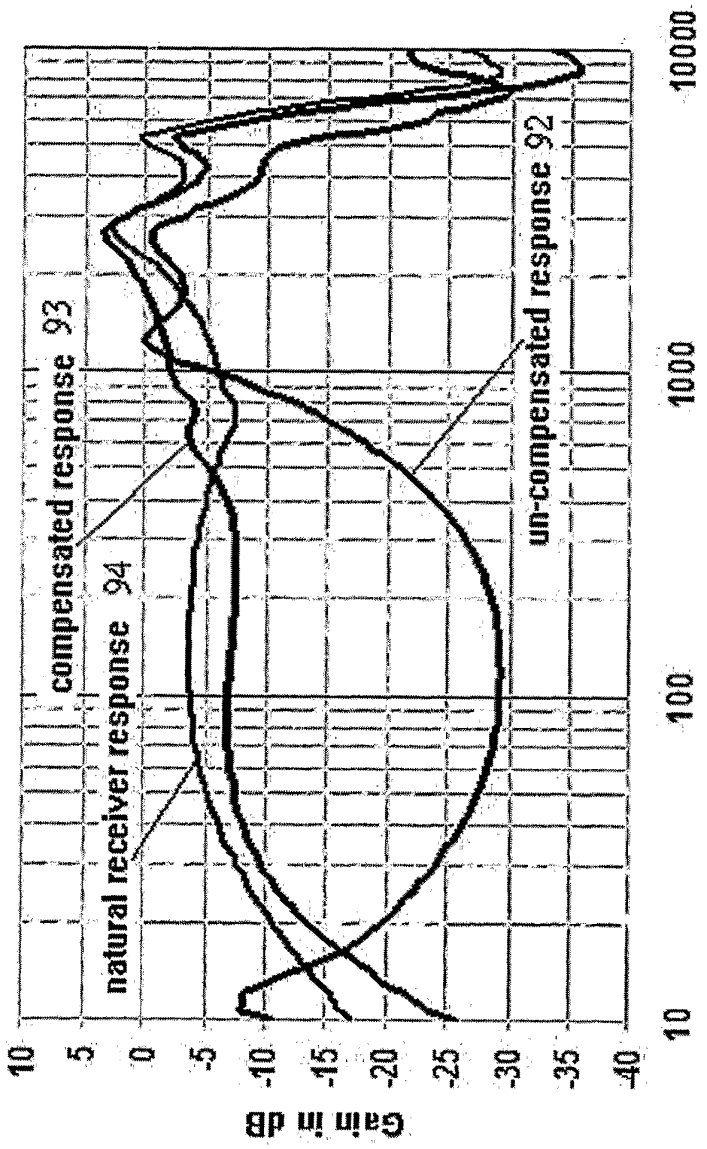

Gain in dB
10
5
0
-5
-10
-15
-20
-25
-30
-35
-40
10
100
1000
10000
Frequency Hz
compensated response 93
natural receiver response 94
un-compensated response 92

Fig. 10

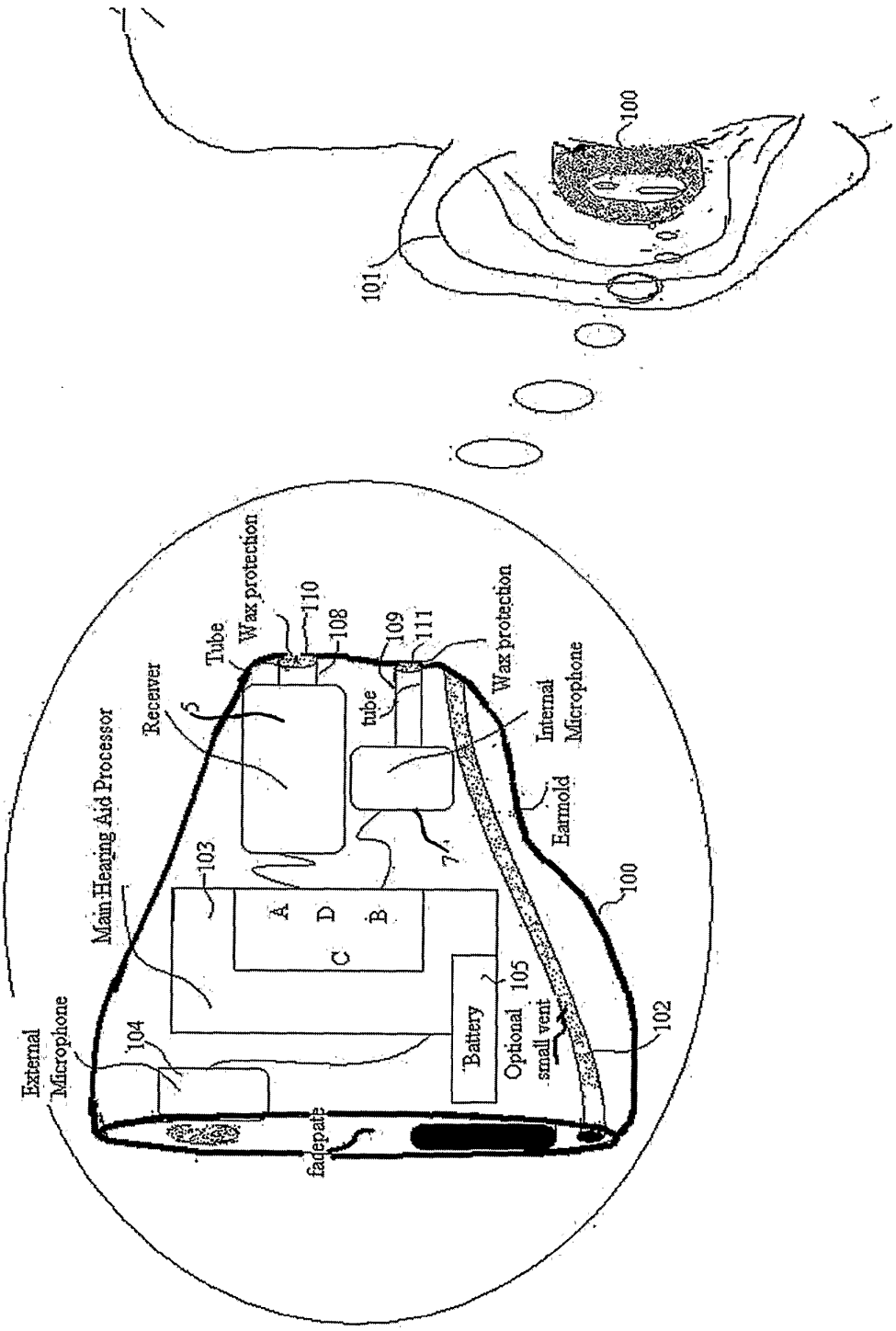

100
101
Wax protection
Tube
110
108
109
111
Wax protection
Receiver
5
tube
Internal Microphone
Earmold
Main Hearing Aid Processor
103
A
D
B
C
100
External Microphone
104
Battery
105
Optional small vent
102
fadepate

Fig. 11

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

A. CLASSIFICATION OF SUBJECT MATTER

Int. Cl. [7]: H04R 29/00; A61B 5/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
WPAT: hearing aid, occlusion, gain, frequency range, sound level, signal strength and similar terms.
USPTO: hearing aid, occlusion, reduce

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 6687377 B2 (VOIX et al.) 3 February 2004 Col. 3 line 33-Col. 6 line 22	1-24
A	WO 2002/085063 A2 (WIDEX A/S et al.) 24 October 2002 Page 4 line 14-Page 10 line 32	1-24
A	US 6766031 B1 (WESTERMANN) 20 July 2004 Col. 2 line 18-Col. 5 line 28	1-24
A	CA 2381516 A1 (GENNUM CORPORATION, CA) 12 October 2002 Page 1 line 20-Page 18 line 2	1-24

[X] Further documents are listed in the continuation of Box C [X] See patent family annex

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 November 2005	9 NOV 2005
Name and mailing address of the ISA/AU AUSTRALIAN PATENT OFFICE PO BOX 200, WODEN ACT 2606, AUSTRALIA E-mail address: pct@ipaustralia.gov.au Facsimile No. (02) 6285 3929	Authorized officer **JUZER KHANBHAI** Telephone No : (02) 6283 2176

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5195139 A (GAUTHIER) 16 March 1993 Col. 1 line 39-Col. 3 line 22	1-24

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

This Annex lists the known "A" publication level patent family members relating to the patent documents cited in the above-mentioned international search report. The Australian Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

Patent Document Cited in Search Report		Patent Family Member					
US	6687377	AU	20394/02	CA	2477010	CN	1483298
		EP	1356255	US	2002076057	WO	0250499
		ZA	200305523				
WO	2002/085063	CA	2442538	EP	1250027	US	2002150269
US	6766031	AU	69189/98	CA	2286037	DK	42297
		EP	0980641	WO	9847318		
CA	2381516	N/A					
US	5195139	AU	20280/92	EP	0584259	WO	9221218

Due to data integration issues this family listing may not include 10 digit Australian applications filed since May 2001.

END OF ANNEX

(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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(19) World Intellectual Property Organization
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(43) International Publication Date
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H04N 7/18 (2006.01) *G06F 19/00* (2006.01)

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PCT/US2007/060620

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(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
60/765,628 6 February 2006 (06.02.2006) US

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(72) Inventor; and

(75) Inventor/Applicant *(for US only)*: **MOBERLY, David** [US/US]; 3899 Magnolia Drive, Palo Alto, California 94306 (US).

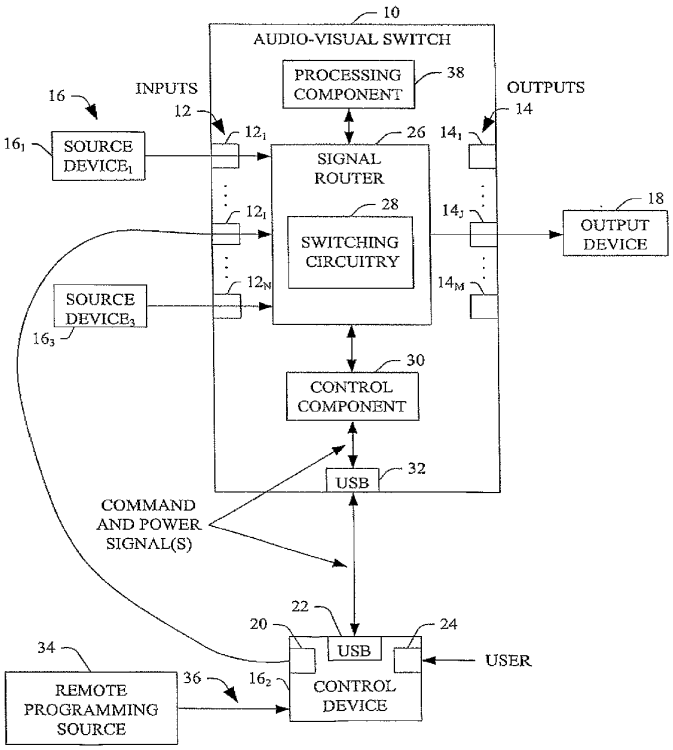
(74) Common Representative: **KONINKLIJKE PHILIPS ELECTRONICS, N.V.**; c/o Douglas B. McKnight, 595 Miner Road, Cleveland, Ohio 44143 (US).

(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LV, LY, MA, MD, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI,

[Continued on next page]

(54) Title: USB-ENABLED AUDIO-VIDEO SWITCH



(57) Abstract: A telemedicine system (8) includes an audio-video switch (10) that receives at least two different signals and conveys at least one of the at least two different signals to an output device (18) that presents it to a user. A control device (162) monitors a position of the audio-video switch (10) and automatically controls the position to select which of the at least two different signals is passed through the audio-video switch (10) to the output device (18) in response to a signal from a remote source (34). Signals received from the remote source (34) include medically related audio video message signals, which are input to the audio-video switch (10), and a control signal that causes the audio-video switch (10) to select between passing the medically related audio-video message signals and other received signals to the output device (18).

WO 2007/092660 A1

FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, **MR,** NE, SN, TD, TG).

Declaration under Rule 4.17:

— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(H))*

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INTERNATIONAL SEARCH REPORT

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A CLASSIFICATION OF SUBJECT MATTER
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B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04N G06F

Documentation searched other than minimum documentation to the extent that such documents are included In the fields searched

Electronic data base consulted during the international search (name of data base and where practical, search terms used)
EPO-Internal , WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No
X	EP 1 126 715 A2 (KATZ RONALD ALAN [US] TELEBUYER LLC [US]) 22 August 2001 (2001-08-22) page 3, paragraph 13 page 9, paragraph 70 page 16, paragraph 134 - page 17, paragraph 151; figures 1-5	1-26
X	US 2005/052527 A1 (REMY CHRISTOPHE [US] ET AL) 10 March 2005 (2005-03-10) page 3, paragraph 30 - paragraph 32; figure 3 page 5, paragraph 41 - paragraph 42; figures 7,8	1-26

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 Apri l 2007	13/04/2007

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P B 5818 Patentlaan 2 NL- 2280 HV Rijεwijk Tel (+31-70) 340-2040,Tx 31 651 epo nl, Fax- (+31-70) 340-3016	Schinnerl , Arnold

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C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation o) document, with Indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 6 717 609 B2 (SUGIMOTO HIDEO [JP] ET AL) 6 April 2004 (2004-04-06) column 7, line 19 - column 8, line 42 abstract; figures 1-5	1-26
X	US 5 553 609 A (CHEN YAOBIN [US] ET AL) 10 September 1996 (1996-09-10) column 9, line 57 - column 12, line 53; figures 5-9	1,23,26
A	EP 1 343 326 A1 (MACROSYSTEM DIGITAL VIDEO A6 [DE]). 10 September 2003 (2003-09-10) abstract; figure 1 column 6, line 22 - line 58	1-26

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/US2007/060620

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 1126715 A2	22-08-2001	NONE	
US 2005052527 A1	10-03-2005	US 2006119701 A1	08-06-2006
US 6717609 B2	06-04-2004	DE 10101064 A1	19-07-2001
		JP 2001197485 A	19-07-2001
		US 2003197781 A1	23-10-2003
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		WO 9624284 A1	15-08-1996
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		DE 50200955 D1	14-10-2004
		US 2003193577 A1	16-10-2003

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INTERNATIONAL SEARCH REPORT

International application No.
PCT/US07/60620

A. CLASSIFICATION OF SUBJECT MATTER
IPC: **H04N 7/18**(2006.01),**9/47**(2006.01)

USPC: 348/61
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 348/61, 65, 74; 600/522

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5,553,609 (CHEN et al.) 10 September 1996, entire document	1-3, 5, 7-11, 13-24
---		---
Y		4, 6, 12, 25-26
Y	US 2008/005421 A1 (CHANG et al.) 03 January 2008; entire document	4, 6, 12, 25-26

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

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Date of the actual completion of the international search
14 July 2008 (14.07.2008)

Date of mailing of the international search report
29 AUG 2008

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Commissioner for Patents
P O **Box 1450**
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(21) International Application Number:
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2006-291843 26 October 2006 (26.10.2006) JP
2006-291844 26 October 2006 (26.10.2006) JP

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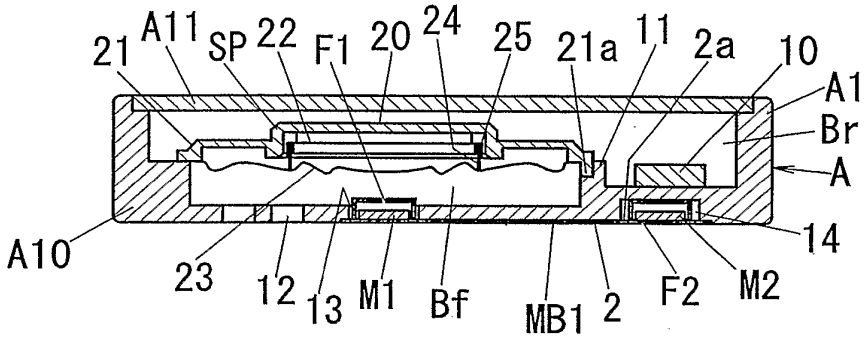
(74) Agents: **NISHIKAWA, Yoshikiyo** et al.; Hokuto Patent Attorneys Office Umeda-Daiichiseimei Bldg. 5th Floor, 12-17, Umeda 1-chome, Kita-ku, Osaka-shi, Osaka 5300001 (JP).

(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

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Published:
— *with international search report*

(54) **Title:** INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME



(57) **Abstract:** A compact intercom device is provided, which efficiently prevents howling phenomenon. This intercom device includes a housing; a speaker accommodated in the housing such that a front air chamber is defined at a front side of the speaker and a rear air chamber is defined as a rear side of the speaker, a first microphone disposed such that its sound receiving surface faces the front air chamber, a second microphone disposed such that its sound receiving surface faces outside of the housing; and a signal processing unit. When an audio output of the speaker is picked up by the second microphone, the signal processing unit removes a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

WO 2008/050583 A1

DESCRIPTION

INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME

TECHNICAL FIELD

The present invention relates to an intercom device suitable for an interphone system or the like, and a wiring system using the same.

BACKGROUND ART

Conventionally, an interphone system has been widely used as short-distance conversation means between separated rooms or between an entrance and a room of a building or a house. A typical interphone system for household use is mainly formed with an indoor intercom device installed on a wall surface of the room, and an outdoor intercom device installed on a wall surface of the entrance, and connected to the indoor intercom device through a transmission line. For example, the outdoor intercom device has a microphone for receiving a visitor's voice, and a speaker for outputting a dweller's voice. By operating the outdoor intercom device, the visitor can have a conversation with the dweller using the indoor intercom device.

In this kind of intercom device, since the speaker is located near the microphone, there is an inconvenience that howling phenomenon easily occurs when the audio output of the speaker is picked up by the microphone. To solve this inconvenience, various measures or methods for preventing the howling phenomenon have been proposed.

For example, Japanese Patent Publication No. 2607257 discloses a loudspeaker communication device **300**, which is suitable for a handsfree telephone. As shown in FIG. 35A, this communication device **300** has a speaker **SP**, and a pair of microphones (**M1**, **M2**). In addition, to prevent the howling phenomenon caused when an audio output of the speaker **SP** is picked up by the microphone (**M1**, **M2**), the communication device **300** also includes a pair of amplifying circuits (**310**, **320**), a level-control amplifying circuit **330**, a

delay circuit **340**, and a differential amplifying circuit **350**, as shown in FIG. 35B. The delay circuit **340** delays an output of the microphone **M1** located closer to the speaker **SP** by a delay time, which is determined according to sound velocity and a difference between a distance **d1** between the microphone **M1** and the speaker **SP** and a distance **d2** between the microphone **M2** and the speaker **SP**. The level-control amplifying circuit **330** controls the output level of the microphone **M2** such that an amplification factor is **d2/d1**. In other words, the output level of the microphone **M2** is controlled so as to be substantially equal to the output level of the microphone **M1** with respect to the signal component corresponding to the audio output of the speaker **SP**. The differential amplifying circuit **350** receives an output of the delay circuit **340** and an output of the level-control amplifying circuit **330**, and provides an output signal corresponding to a difference between these outputs.

According to this communication device, the signal component corresponding to the audio output of the speaker **SP** is cancelled from the output of the microphone (**M1**, **M2**) by the differential amplifying circuit **350**. Therefore, it is possible to achieve a comfortable communication, while preventing the howling phenomenon.

However, as shown in FIG. 35A, since each of the microphones (M1, M2) is spaced from the speaker SP by a relatively large distance, there is another problem that the device itself becomes large in size. In addition, since the intercom device is usually installed on the wall surface or the like, it is strongly desired that the intercom device has a thin thickness. On the other hand, when the speaker is accommodated in an excessively thin housing, a volume of a rear air chamber, which is defined as a space of a rear side of the speaker, is reduced, so that a radiating sound pressure of the speaker lowers and a minimum resonance frequency of the speaker shifts to a higher frequency level. As a result, there is a fear that deteriorations in sound quality and electro-acoustic conversion efficiency of the speaker occur.

Thus, the conventional intercom device still has plenty of room for

improvement in downsizing of the device itself, sound quality, electro-acoustic conversion efficiency as well as the prevention of howling phenomenon.

SUMMARY OF THE INVENTION

Therefore, a primary concern of the present invention is to provide a compact intercom device, which has the capability of preventing howling phenomenon, and achieving an improvement in sound quality and efficiency.

That is, the intercom device of the present invention is characterized by comprising:

a housing;

a speaker accommodated in the housing such that a front air chamber is defined as a space enclosed by a front surface of the speaker and an inner surface of the housing, and a rear air chamber is defined as a space enclosed by a rear surface of the speaker and an inner surface of the housing;

a first microphone disposed such that its sound receiving surface faces the front air chamber;

a second microphone disposed such that its sound receiving surface faces outside of the housing; and

a signal processing unit configured to, when an audio output of the speaker is picked up by the second microphone, remove a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

According to the invention, since the first microphone easily and efficiently collects the audio information output from the speaker, howling phenomenon can be efficiently prevented by the signal processing unit. In addition, a distance between the first microphone and the speaker becomes short, thereby allowing downsizing of the intercom device. Furthermore, since the sound receiving surface of the second microphone faces the outside of the housing, an acoustic coupling between the speaker and the second microphone reduces, and the second microphone becomes hard to pick up the audio output of the

speaker. As a result, the second microphone can be located closer to the speaker. Thus, it is possible to provide a compact intercom device capable of preventing the howling phenomenon.

From the viewpoint of more efficiently preventing the howling phenomenon, it is preferred that the rear air chamber is the space hermetically-sealed by the rear surface of the speaker and the inner surface of the housing. In addition, it is preferred that the first microphone is arranged such that the sound receiving surface of the first microphone faces a diaphragm of the speaker. Moreover, when the first microphone and the second microphone are mounted on a same circuit board, it is preferred that the circuit board is mounted on an outside surface of the housing such that the sound receiving surface of the first microphone faces the front air chamber and the sound receiving surface of the second microphone faces the outside of the housing.

On the other hand, from the viewpoint of further downsizing the intercom device, the second microphone is preferably located adjacent to the speaker. In addition, it is preferred that the first and second microphones are mounted on a single board.

From the viewpoint of improving sound quality and electro-acoustic conversion efficiency of the speaker, it is preferred that the intercom device has an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end. Alternatively, the intercom device may have a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end. In addition, it is preferred that the acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of the speaker is needed.

Furthermore, it is preferred that the acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant frequency of the speaker that is supposed to be mounted on an infinite baffle and a lowest

resonant frequency of the speaker mounted on the rear air chamber with the absence of the acoustic tube. The acoustic tube can be formed along an inner surface of the rear air chamber. If necessary, a sound absorbing material may be disposed at the opening of the acoustic tube.

A further concern of the present invention is to provide a wiring system using the intercom device described above. That is, the wiring system comprises at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of the building structure, and connected to the transmission line. The intercom device has a first connector detachably connectable to a second connector formed in each of the base units. This wiring system is characterized in that when the first connector is connected to the second connector of one of the base units, the intercom device functions such that the information signals provided through the transmission line are output as audio information from the speaker unit, and an output of the signal processing unit is sent out through the transmission line.

By the way, in the intercom device installed on the wall surface of the house, one provided with a liquid crystal display or the like for displaying not only the audio information but also visual information such as images comes into practical use recently. Once such an intercom device is installed on the wall surface, not only electric wiring work but also mounting/repair work to the wall surface are required in a case where a layout of the intercom device has to be changed. However, it is not easy for a general user himself/herself to do the work. In addition, a function of the conventional intercom device is completed by itself, and a new-type intercom device is required in a case where it is desired to add another function. In such a case, not only a purchase of the new-type intercom device but also the above-described mounting/repair work becomes necessary, so that this is a great burden in cost for the user. The wiring system of the present invention is effect. According to the wiring system of the present

invention, since the intercom device is detachably connectable to the base unit, there advantages that the wiring system allows great flexibility for the change of layout of the intercom device, and makes it easy to replace the intercom device without cumbersome repairing work. Thus, the present invention provides the wiring system capable of solving the problems described above.

As a preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units, the function unit having a third connector detachably connectable to one of the second connector of the base unit, and an extended connector formed in the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, or when the first connector of the intercom device is connected to the second connector of the base unit, and the third connector of the function unit is connected to the extended connector of the intercom device, the function unit provides at least one of functions for supplying electric power provided through the transmission line, outputting information signals provided through the transmission line, and sending out information signals through the transmission line. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

As another preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units. The function unit has a third connector detachably connectable to the second connector of the base unit and an extended connector detachably connectable to the first connector of the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, and the extended connector of the function unit is connected to the first connector of the

intercom device, the function unit provides at least one of functions for supplying electric power provided from the transmission line through the base unit, outputting information signals provided from the transmission line through the base unit, and sending out information signals to the transmission line through the base unit, and the intercom device operates such that the information signals provided from the transmission line through the base unit and the function unit are output as audio information from the speaker unit, and the output of the signal processing unit is sent out to the transmission line through the base unit and the function unit. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

In the wiring system described above, it is preferred that the first and second connectors provide an electric power transmission between the base unit and the intercom device by means of electromagnetic coupling, and/or the first and second connectors provide a signal transmission between the base unit and the intercom device by means of optical coupling.

Further characteristics of the present invention and advantages brought thereby will be clearly understood from the best mode for carrying out the invention described below.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a perspective view of an intercom device according to a first embodiment of the present invention;

FIG. 2 is a cross-sectional view of the intercom device;

FIG. 3 is a circuit diagram of a voice processing unit of the intercom device;

FIG. 4 is a cross-sectional view of a microphone board of the intercom device;

FIG. 5 is a cross-sectional view of a bare chip employed in the microphone board;

FIG. 6A is a simplified plan view of the microphone board, and FIG. 6B is a simplified circuit diagram of the microphone board;

FIG. 7 is a circuit diagram of an impedance converting circuit of the microphone board;

FIG. 8 is a circuit diagram of a signal processing unit of the intercom device;

FIG. 9A and FIG. 9B are diagrams showing signal waveforms in the signal processing unit;

FIG. 10A and FIG. 10B are diagrams showing further signal waveforms in the signal processing unit;

FIG. 11A and FIG. 11B are diagrams showing another signal waveforms in the signal processing unit;

FIGS. 12A to 12C are diagrams showing still another signal waveforms in the signal processing unit;

FIG. 13 is a cross-sectional view of an intercom device according to a second embodiment of the present invention;

FIGS. 14A to 14C are cross-sectional and plan views of another housing of the intercom device according to the second embodiment;

FIG. 15 is a graph showing frequency characteristics of the cancelled amount of voice components of the speaker;

FIG. 16 is a graph showing frequency characteristics of the radiating sound pressure of the speaker;

FIG. 17 is a cross-sectional view of an intercom device using an ideal baffle plate;

FIG. 18 is a graph showing sound pressure characteristics of the intercom device having an acoustic tube and the intercom device having no acoustic tube;

FIG. 19 is a simplified cross-sectional view showing a modification of the housing having the acoustic tube;

FIG. 20 is a simplified cross-sectional view of a housing having a plurality of acoustic tubes;

FIG. 21 is a plan view of a housing having a sound absorbing material;

FIG. 22 is a schematic diagram of a wiring system using an intercom device according to a third embodiment of the present invention;

FIG. 23 is a schematic circuit diagram of a base unit of the wiring system;

FIG. 24 is a rear perspective view of the base unit;

FIG. 25A is a perspective view showing a gate housing and a main housing of the base unit, and FIG. 25B is a plan view of a module port of the gate housing;

FIG. 26 is a plan view of an attachment plate for mounting the base unit on a switch box;

FIG. 27 is a schematic circuit diagram of the function unit;

FIG. 28 is a perspective view of the intercom device for the wiring system;

FIG. 29 is a schematic circuit diagram of the intercom device for the wiring system;

FIG. 30 is a perspective view of the intercom device connected to the function unit or the base unit;

FIG. 31 is an exploded perspective view of the base unit;

FIG. 32A is a front view of the function unit, FIG. 31B is a side view of the function unit, FIG. 32C is an exploded side view of the function unit, and FIG. 32D is a perspective view of a joining member;

FIG. 33 is a perspective view illustrating how to mechanically couple the function unit to the base unit;

FIG. 34 is a schematic circuit diagram of an intercom device for a power line carrier type wiring system; and

FIG. 35A is a schematic perspective view of a conventional loudspeaker communication device, and FIG. 35B is a partial block diagram of the loudspeaker communication device.

BEST MODE FOR CARRYING OUT THE INVENTION

Referring to the attached drawings, an intercom device of the present invention and a wiring system using the intercom device are explained in detail

according to preferred embodiments.

(First Embodiment)

As shown in FIGS. 1 and 2, an intercom device **A** according to a first embodiment of the present invention includes a housing **A1**, a speaker **SP**, a microphone board **MB1**, and a voice processing unit **10**. The housing **A1** consists of a body **A10** having an opening at its back face, and a cover **A11** for closing the opening of the body **A10**, and accommodates the speaker **SP** and the voice processing unit **10** therein. The housing **A10** has a plurality of sound holes **12** formed in a front face of the body **A10**. The speaker **SP** accommodated in the housing **A1** provides a voice output through the sound holes **12** to outside of the housing **A1**. The microphone board **MB1** has a sound hole **F2** that picks up a voice emitted from a user of the intercom device **A**. In operation, when a conversation switch **SW1** is operated to the ON position, the intercom device **A** sends the picked-up voice to another intercom device **A** and receives signals from another intercom device **A** via an information line **Ls**.

As shown in FIG. 2, the speaker **SP** includes a cylindrical yoke **20** that is open at one end, and a round support **21** that extends radially outward from the open end of the yoke **20**. The yoke **20** is formed of an iron-containing material, such as a cold rolling steel plate (SPCC, SPCEN) or an electromagnetic soft iron (SUY), and has a thickness of about 0.8 mm. A columnar permanent magnet **22** (having a residual flux density of, for example, 1.39 T to 1.43 T) formed of, for example, neodymium is placed in the yoke **20**. An edge surface of the round support **21** is fixed on an edge portion on an outer peripheral side of a dome-shaped diaphragm **23**.

The diaphragm **23** is formed of a thermoplastic material, such as PET (PolyEthylene Terephthalate) or PEI (Polyetherimide), and has a thickness of, for example, 12 μm to 50 μm. A cylindrical bobbin **24** is fixed to the back of the diaphragm **23**, and a voice coil **25** is provided at the rear end of the bobbin **24**. The voice coil **25** is located at the open end of the yoke **20**, and the bobbin **24**

and the voice coil **25** are allowed to freely move back and forth (in the vertical direction in FIG. 2) in the vicinity of the open end of the yoke **20**. The voice coil **25** is formed by winding a polyurethane copper wire (having a diameter of 0.05 mm, for example) around a paper sleeve of craft paper.

When the polyurethane copper wire of the voice coil **25** receives a voice signal, an electromagnetic power is generated in the voice coil **25** by electric current of the voice signal and a magnetic field of the permanent magnet **22**, thereby to vibrate the bobbin **24**, along with the diaphragm **23**, forward and backward. As a result, a sound corresponding to the voice signal is emitted from the diaphragm **23**. That is to say, the speaker **SP** of this embodiment is an electrodynamic speaker. The speaker has, for example, a diameter of 20 mm to 25 mm, and a thickness of about 4.5 mm.

A rib **11** is formed on an inner surface of the front wall of the housing **A1**, which is opposed to the diaphragm **23**, and is in contact with an end face of a convex portion **21a** that protrudes frontward from an outer peripheral portion of the round support **21**. Thus, the speaker **SP** is fixed to the housing **A1** such that the diaphragm **23** faces the inner surface of the front wall of the housing **A1**.

With the speaker **SP** thus fixed within the housing **A1**, a front air chamber **Bf** is formed as a space enclosed by the inner surface of the front wall of the housing **A1** and a front surface (on which the diaphragm **23** is located) of the speaker **SP**, and a rear air chamber **Br** is formed as a space enclosed by an inner surface of the rear and side walls of the housing **A1** and the rear surface (on which the yoke **20** is located) of the speaker **SP**. The front air chamber **Bf** communicates with outside of the housing **A1** through the sound holes **12** formed in the front wall of the housing **A1**. The rear air chamber **Br** is insulated from (or does not communicate with) the front air chamber **Bf** since the end portion of the support **21** and the rib **11** are attached firmly to each other. Furthermore, the cover **A11** is tightly fitted in the opening formed in the rear wall of the body **A10**, so that the rear air chamber **Br** provides a

hermetically-sealed space that is insulated also from the outside of the housing **A1**. A gasket may be provided between the end portion (or the convex portion **21a**) of the support **21** and the rib **11** so as to increase the degree of adhesion between the support **21** and the rib **11**.

As shown in FIG. 3, the voice processing unit **10** is formed by an integral circuit (IC), which includes a communication unit **10a**, echo canceling units (**10b**, **10c**), an amplification unit **10d** and a signal processing unit **10e**. In operation, the communication unit **10a** receives a voice signal transmitted from an intercom device **A** installed, for example, in another room through the information line **Ls**. The received voice signal is transmitted through the echo canceling unit **10b** to the amplification unit **10d**, in which the signal is amplified, and then output from the speaker **SP**. When the conversation switch **SW1** is operated to the ON position, the voice communication function of the intercom device **A** becomes available. In this condition, voice signals respectively received from a microphone **M1** (first microphone) and a microphone **M2** (second microphone) on the microphone board **MB1** are processed by the signal processing unit **10e**. In the signal processing unit **10e**, each received signal is processed in the manner as described later, and the resulting signal passes through the echo canceling unit **10c**. Then, upon reception of the resulting signal from the echo canceling unit **10c**, the communication unit **10a** transmits the resulting signal through the information line **Ls** to the intercom device **A** installed in, for example, another room. In this manner, the intercom device **A** serves as an interphone that permits two-way voice communications between the rooms. An electric power for the intercom device **A** may be supplied from a wall socket disposed in the vicinity of the location where the intercom device **A** is installed. Alternatively, the electric power may be supplied through the information line **Ls**, or may be supplied from an internal battery accommodated in the intercom device **A**.

As shown in FIG. 4, the microphone board **MB1** has a module board **2** and the microphones **M1** and **M2**, which are mounted on a (top) surface **2a** of the

module board **2**. The microphone **M1** is mainly formed with a bare chip **BC1**, an IC chip **Ka1**, and a shield case **SC1**. Each of the bare chip **BC1**, the IC chip **Ka1**, and a wiring pattern (not shown) on the module board **2** are connected to each other with wires **W** (wire bonding). The shield case **SC1** covers the bare chip **BC1** and the IC chip **Ka1**. The microphone **M2** has the same construction as that of the microphone **M1**. That is to say, the microphone **M2** is formed with a bare chip **BC2**, an IC chip **Ka2**, and a shield case **SC2** mounted on the module board **2** that covers the bare chip **BC2** and the IC chip **Ka2**. The microphones (**M1**, **M2**) are preferred to be formed on a silicon substrate by a micro structure manufacturing process. That is, it is preferred that the microphones (**M1**, **M2**) are IC chips of Micro Electro Mechanical System (MEMS).

Each of the bare chips **BC1** and **BC2**, which will be generally referred to as "bare chip **BC**", provides a condenser-type silicon microphone, as shown in FIG. 5. More specifically, the bare chip BC includes a silicon thin film **1d** that is formed on a (top) surface of a silicon substrate **1b** so as to cover a cavity **1c** formed through the silicon substrate **1b**. An electrode **1f** is formed above the silicon thin film **1d** with an air gap **1e** interposed therebetween, and two pads **1g** are provided on the surface of the silicon substrate **1b** for producing an output in the form of a voice signal. When the silicon thin film **1d** is vibrated by a sound wave from outside, such as a human voice, a capacitance between the silicon thin film **1d** and the electrode **1f** changes, namely, the amount of electric charge changes. With the change in the electric charge amount, electric current proportional to the received sound wave flows from the two pads **1g**. Each of these bare chips **BC1** and **BC2** is fixed by die-bonding the silicon substrate **1b** on the module board **2**, as shown in FIG. 4. The silicon thin film **1d** of the bare chip **BC2** faces the sound hole **F2** formed in the module board **2**.

As shown in FIG. 4, a sound receiving surface of the microphone **M1** is positioned on a surface having the sound hole **F1** of the shield case **SC1**. On the other hand, a sound receiving surface of the microphone **M2** is positioned

on a surface having the sound hole **F2** of the module board **2**. Thus, the sound receiving surface of the microphone **M1** is to receive sound from an upper side of FIG. 4, and the sound receiving surface of the microphone **M2** is to receive sound from an opposite side, i.e., the lower side of FIG. 4. In brief, as shown in FIG. 2, the module board **2** is mounted to the housing **A1** such that the sound receiving surface of the microphone **M1** faces the diaphragm **23** of the speaker **SP**, and the sound receiving surface of the microphone **M2** faces outside of the housing **A1**. Since both of the microphones (**M1**, **M2**) are mounted on the same surface **2a** of the module board **2**, the thickness of the microphone board **MB1** can be advantageously reduced.

FIG. 6A is a simplified plan view of the module board **2**. The module board **2** has a substantially T-shape, which is composed of a rectangular portion **2f**, on which the microphone **M1** is mounted, a rectangular portion **2g**, on which the microphone **M2** is mounted, and a connecting portion **2h** that connects the rectangular portions **2f** and **2g**. The rectangular portion **2g** is formed to be larger than the rectangular portion **2f**. The rectangular portion **2g** has a negative power supply pad **P1**, a positive power supply pad **P2**, and output pads (**P3**, **P4**) on an edge portion of the rectangular portion **2g** such that each of the pads (**P1**, **P2**, **P3**, **P4**) forms a line along an end face of the edge portion of the rectangular portion **2g**.

As shown in FIG. 6B, the negative power supply pad **P1** is connected to a negative terminal of an external power supply, and the positive power supply pad **P2** is connected to a positive terminal of the external power supply. The electric power received from the pads (**P1**, **P2**) is supplied to the microphones (**M1**, **M2**) through the wiring patterns on the module board **2**. A voice signal corresponding to the voice picked up by the microphone **M1** is output from the output pad **P3** via the wiring pattern on the module board **2**. On the other hand, a voice signal corresponding to the voice picked up by the microphone **M2** is output from the output pad **P4** via the wiring pattern on the module board **2**. The negative power supply pad **P1** also serves as a ground pad that establishes

a ground for the voice signals delivered from the output pads (**P3**, **P4**).

As described above, electric powers needed to activate the microphones (**M1**, **M2**) are supplied from the power supply via the common negative and positive power supply pads (**P1**, **P2**). The negative power supply pad **P1** also serves as the ground pad for the outputs of microphones (**M1**, **M2**). Therefore, the number of pads used in the microphone board **MB1** can be reduced, and the construction of the microphone board **MB1** can be simplified.

Next, an operation of the microphone board MB1 will be explained. First, electric currents flowing from the bare chips **BC1** and **BC2** corresponding to the received sound signal are subjected to impedance-conversion, and converted to voltage signals by the IC chips **Ka1** and **Ka2**. Then, the voltage signals are output as voice signals from the output pads (**P3**, **P4**), respectively.

For example, each of the IC chip **Ka1** and the IC chip **Ka2**, which will be generally referred to as "IC chip **Ka**", has a circuit configuration, as shown in FIG. 7. The IC chip **Ka** has a constant-voltage circuit **Kb** formed of a chip IC that converts a power supply voltage +V (e.g., 5V) supplied from the power supply pads (**P1**, **P2**) to a constant voltage Vr (e.g., 12V). The constant-voltage circuit **Kb** applies the constant voltage Vr to a series circuit of a resistance **R11** and the bare chip **BC**. A connection midpoint of the resistance **R11** and the bare chip **BC** is connected to a gate terminal of a junction-type J-FET **S11** through a capacitor **C11**. A drain terminal of the J-FET **S11** is connected to an operating voltage +V, and a source terminal is connected through a resistance **R12** to the negative terminal of the power supply. Herein, the J-FET **S11** is intended for converting electrical impedance, and the voltage of the source terminal of the J-FET **S11** produces an output as the voice signal. It is to be understood that the impedance converting circuit of the IC chip **Ka** is not limited to the above-described configuration, and may be replaced by a circuit having a function of a source follower circuit by an operation amplifier. For example, an amplifying circuit for the voice signal may be provided in the IC chip **Ka**, if necessary.

In the microphone board **MB1**, on which signal transmission and supply of electric power are performed via the wiring patterns on the module board **2** as described above, the signal lines and the power-supply lines can be arranged with high efficiency, and attached to the outer surface of the housing **A1**, as shown in FIG. 1. In this embodiment, as shown in FIG. 1, the surface **2a** of the module board **2** is attached along an outer surface of the front wall of the housing **A1**, and the microphone **M1** is inserted in an opening **13** formed in the front wall of the housing **A1** such that the sound receiving surface of the microphone **M1** faces the front air chamber **Bf**. The diaphragm **23** faces the microphone **M1** through the sound hole **F1** formed in the shield case **SC1**. With this arrangement, the microphone **M1** has a high directivity to the voice emitted from the speaker **SP** via the sound hole **F1**, so that the voice emitted from the speaker **SP** can be efficiently picked up by the microphone **M1**. The microphone **M2** fits in a recess **14** formed in the front wall of the housing **A1**, and faces the outside (or the front) of the housing **A1** through the sound hole **F2** formed in the module board **2**. In this regard, a direction where the second microphone **M2** receives a voice from the outside of the housing **A1** is substantially equal to the direction of emitting a voice from the speaker **SP**. Thus, the microphone **M2** has a high directivity to the voice of a talker who is located in front of the intercom device **A**. When **X1**, **X2** represent distances from the center of the speaker **SP** to the centers of microphones (**M1**, **M2**), respectively, the relationship between the distances (**X1**, **X2**) is expressed as **X1** $<$**X2**.

Since the rear air chamber **Br** facing the rear surface of the speaker **SP** is hermetically-sealed in the housing **A1** as shown in FIG. 2, the voice emitted from the rear surface of the speaker **SP** (rear surface of the diaphragm **23**) is less likely to leak from the rear air chamber **Br**, so that an acoustic coupling between the speaker **SP** and the microphone **M2** is lowered. BY the way, the voice emitted from the rear surface of the speaker **SP** is reversed in phase with respect to the voice emitted from the front surface of the speaker **SP** (front surface of the

diaphragm **23**). Therefore, if the voice emitted from the rear surface of the speaker **SP** goes around to the front of the speaker **SP**, these voices emitted from the front and rear surfaces of the speaker **SP** are cancelled with each other. As a result, the radiating sound pressure of the speaker **SP** is lowered, and it becomes difficult for the talker located in front of the intercom device **A** to hear the voice (sound) emitted from the speaker **SP**. However, since the intercom device of this embodiment is formed to reduce the leaking of the voice emitted from the rear surface of the speaker **SP** to the outside of the housing **A1**, as described above, it is possible to prevent such a reduction in the radiating sound pressure, which is caused by the voice emitted from the rear surface of the speaker **SP** going around to the front of the speaker **SP**.

As shown in FIG. 2, since the recess **14**, in which the microphone **M2** is inserted, is an insulated space not communicated with the rear air chamber **Br**, the microphone **M2** is even less likely to pick up the voice (or sound) emitted from the speaker **SP**. Therefore, the acoustic coupling between the speaker **SP** and the microphone **M2** can be further reduced. With the above arrangement, the microphones (**M1**, **M2**) are adapted to separately pick up the voice emitted from the speaker **SP** and the voice of (generated by) the talker, respectively.

Furthermore, if the microphone board **MB1** is placed in the housing **A1**, it is difficult to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**. In this embodiment, however, the microphone board **MB1** is attached to the outer surface of the housing **A1**. Therefore, it is possible to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**.

As described below, the intercom device of this embodiment has a configuration to prevent a howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**.

As shown in FIG. 8, the signal processing unit **10e** of the voice processing unit **10** includes an amplifying circuit **30** for amplifying the output of the microphone **M1**, a band-pass filter **31** for removing noises of frequencies other

than a voice-band (300 to 4000 Hz) from the output of the amplifying circuit **30**, and a delaying circuit **32** for delaying the output of the band-pass filter **31**. The signal processing unit **10e** further includes an amplifying circuit **33** for amplifying the output of the microphone **M2** while reversing the phase of the output of the microphone **M2** by 180 degrees, a band-pass filter **34** for removing noises of frequencies other than the voice-band (300 to 4000 Hz) from the output of the amplifying circuit **33**, and an adder circuit **35** for adding the outputs of the delaying circuit **32** and the band-pass filter **34**.

Referring to FIG. 8 to FIG. 12C, operations of the signal processing unit **10e** will be explained below. First, an operation of the signal processing unit **10e** when the voice emitted from the speaker **SP** is picked up by the microphones (**M1**, **M2**) will be explained. FIGS. 9A and 9B, 10A and 10B, 11A and 11B, and 12A to 12C show voice signal waveforms that appear at respective units of the signal processing unit **10e** when the microphones (**M1**, **M2**) respectively pick up the voice emitted from the speaker **SP**. As mentioned above, the distance **X1** from the center of the speaker **SP** to the center of the microphone **M1** is smaller than the distance **X2** from the center of the speaker **SP** to the center of the microphone **M2**. Accordingly, as shown in FIGS. 9A and 9B, when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**, the amplitude of the output **Y21** of the microphone **M2** is smaller than that of the output **Y11** of the microphone **M1** due to the difference between the distances (**X1**, **X2**). In addition, the phase of the output **Y21** of the microphone **M2** is delayed with respect to that of the output **Y11** of the microphone **M1** by a delay time **Td** (=(**X2**-**X1**)/**Vs**, where **Vs** is sonic velocity), which is proportional to the difference between the distances (**X2**-**X1**).

The amplifying circuit **30** amplifies the output **Y11** to produce an output **Y12**, and the amplifying circuit **33** amplifies the output **Y21** while reversing the phase thereof by 180 degrees, to produce an output **Y22**. At this time, each of the amplifying circuits (**30**, **33**) performs level adjustment based on the difference between the distances (**X2**-**X1**), so that the output levels of the

microphones (**M1**, **M2**) relative to the voice emitted from the speaker **SP** become equal to each other, as shown in FIGS. 10A and 10B. In this embodiment, since an amplitude rate of the amplifying circuit **30** is set to substantially 1, the amplifying circuit **30** may be eliminated from the signal processing unit **10e**.

As shown in FIGS. 11A and 11B, the band-pass filters (**31**, **34**) remove frequencies other than the voice band from the outputs **Y12** and **Y22**, to produce outputs **Y13** and **Y23**, respectively.

The delaying circuit **32** can be formed by, for example, a time delay device or a CR phase delaying circuit. The delaying circuit **32** delays the output of the microphone **M1**, which is located closer to the speaker **SP**, by the above-described delay time **Td**, thereby producing an output **Y14** whose phase coincides with the phase of the output **Y23** of the band-pass filter **34**. Thus, it is possible to reduce noises contained in a voice signal to be transmitted.

Through the amplifying and delaying processes described above, a voice component emitted from the speaker **SP**, which is contained in the output **Y14**, and the voice component emitted from the speaker **SP**, which is contained in the output **Y23**, have the same amplitude and the reversed phase relative to each other, as shown in FIG. 12A and FIG. 12B. The adder circuit **35** adds the output **Y14** and the output **Y23** to cancel the voice component of the speaker **SP** from the output of the microphone **M2**. Thus, it is possible to produce an output **Ya** shown in FIG. 12C, in which the voice component from the speaker **SP** is removed or reduced. As described above, the signal processing unit **10e** removes the voice component of the speaker **SP** from the output signal **Y21** of the microphone **M2** by using the output signal **Y11** of the microphone **M1**.

On the other hand, an operation of the signal processing unit **10e** when the voice emitted from the talker **H** in front of the intercom device **A** is picked up by the microphones (**M1**, **M2**) will be described below. Since the microphone **M2** has a high directivity to the voice from the talker **H** located in front of the intercom device **A**, the amplitude of the output **Y21** of the microphone **M2** is larger than that of the output **Y11** of the microphone **M1**. Furthermore, since

the signal from the microphone **M1** is not amplified while the signal from the microphone **M2** is amplified in the amplifying circuit **33**, the voice component from the talker **H** contained in the output **Y23** is sufficiently larger than the voice component from the talker **H** contained in the output **Y14**. Thus, even after the above-described adding process is performed in the adder circuit **35**, the signal corresponding to the voice generated by the talker **H** remains in the output **Ya** with a sufficiently large amplitude.

In the manner described above, the voice component from the speaker **SP** is reduced in the output **Ya** of the adder circuit **35**, whereas the voice component generated by the talker **H** in front of the intercom device **A** remains in the output **Ya**. Thus, a relative difference in the output **Ya** between the voice component from the talker **H**, which is desired to be saved, and the voice component from the speaker **SP**, which is desired to be reduced, becomes larger. That is to say, even when the voice from the talker **H** and the voice from the speaker **SP** are simultaneously generated, only the voice component from the speaker **SP** is reduced while the voice component from the talker **H** remains sufficiently large. Accordingly, it is possible to prevent the howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice output of the speaker **SP**.

Since the microphone **M1** is disposed in the front air chamber **Bf** so as to face the diaphragm **23**, as shown in FIG. 2, the microphone **M1** efficiently picks up the voice emitted from the speaker **SP**. Therefore, the signal processing unit **10e** can efficiently achieve the howling phenomenon prevention function. Furthermore, since the sound receiving surface of the microphone **M2** faces outside of the housing **A1**, and the output direction of the speaker **SP** and the directivity of the microphone **M2** substantially conform to each other, the acoustic coupling between the speaker **SP** and the microphone **M2** is reduced. Thus, the microphone **M2** is less likely to pick up the voice emitted from the speaker **SP**, and can be placed close to the speaker **SP**, that is to say, close to the front air chamber **Bf**, which makes it possible to downsize the intercom device **A**.

On the other hand, the signal processing unit **10e** transmits the voice signal to the echo canceling unit **10c** through a digital-analog converter (not shown), and the echo canceling unit (**10b**, **10c**) shown in FIG. 3 perform the process as described below to further efficiently prevent the howling phenomenon.

First, the echo canceling unit **10c** retrieves the output of the echo canceling unit **10b** as a reference signal, and performs a suitable computation on the output of the signal processing unit **10e**, thereby canceling a voice signal, which goes around from the speaker **SP** to the microphones (**M1**, **M2**). On the other hand, the echo canceling unit **10b** also retrieves the output of the echo canceling unit **10c** as the reference signal, and performs a suitable computation on the output of the communication unit **10a**, thereby canceling a voice signal, which goes around from a speaker of an intercom device installed in another room to its microphones.

More specifically, the echo canceling units (**10b**, **10c**) control the amount of loss at variable loss means (not shown) provided in a loop circuit consisting of the speaker **SP**, the microphones (**M1**, **M2**), the signal processing unit **10e**, the echo canceling unit **10c**, the communication unit **10a**, the echo canceling unit **10b**, the amplifying unit **10d** and the speaker **SP**, so as to prevent the howling phenomenon by making the loop gain less than 1. Herein, one of the transmitted voice signal and the received voice signal, which has a smaller signal level, is regarded as not important, and a transmission loss of a variable loss circuit, which is inserted in a transmission path of the signal having the smaller level, is increased. In this embodiment, the intercom devices transmit signals via the information line **Ls**. However, the transmission of signals is not limited to a wired line. A non-wired transmission manner such as radio wave, optical communication system, and infrared radiation may be alternatively employed.

(Second Embodiment)

As shown in FIG. 13, an intercom device **A'** according to a second

embodiment of the present invention includes an acoustic tube **40**, which is provided in the rear air chamber **Br** having a small volume, so as to improve the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. The intercom device **A'** according to the second embodiment will be explained below.

The acoustic tube **40** has an opening at its one end and a closed end at the other end, and is provided in the rear air chamber **Br** to be along an inner wall surface(s) of the rear air chamber **Br**. By setting the resonance frequency of the acoustic tube **40** to an appropriate level, the lowest resonance frequency **fo** of the speaker **SP**, which is provided in the intercom device **A'**, is shifted to a lower frequency level, and the sound pressure level of the speaker **SP** increases, resulting in improvements of the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. It is noted that except for the sound tube **40**, the construction of the intercom device **A'** shown in FIG. 13 is the same as that of the intercom device **A** shown in FIG. 2. Accordingly, details of the intercom device **A'** will be omitted herein.

The lowest resonance frequency **fo** of the speaker **SP** is generally determined by an equivalent mass (diaphragm, coil, additional mass of air) **Mo** of a vibration system of the speaker, stiffness **So** of an edge or the like supporting the vibration system, and stiffness **Sc** of air in the rear air chamber **Br**. That is, it is represented by an equation:

$$fo = \{1/(2\pi)\} \cdot \sqrt{\{(So+Sc)/Mo\}}$$

FIG. 14A through FIG. 14C show another housing **A2** of the intercom device **A'** according to a modification of the present embodiment. The housing **A2** accommodates the acoustic tube **40** therein. Effects of the sound tube **40** are explained below referring to the housing **A2** as an example. The housing **A2** essentially consists of a body **A20** having opening at its rear surface, and a cover **A21** for closing the opening of the body **A20**. In FIG. 14B, only the body **A20** is illustrated. As in the intercom device **A** of the first embodiment, the housing **A2** is capable of accommodating a speaker **SP**, a microphone board **MB1**, a

conversation switch **SW1**, and a voice processing unit **10**. The acoustic tube **40** is provided in the body **A20**.

The housing **A2** has two spaces **A2a** and **A2b** therein, which are insulated from each other. The space **A2a** has a lateral dimension of 30 mm, a vertical dimension of 40 mm, and a thickness of 8 mm, to accommodate the speaker **SP** therein. In the space **A2a**, columnar bosses **51** are formed at regularly spaced four locations on the inner surface of the front wall of the body **A20**, and screw holes **52** are respectively formed in the bosses **51**. The speaker **SP** is fixed to the inner surface of the front wall of the body **A20** by using mounting screws, which are thread engageable with the screw holes **52**. With the construction described above, a front air chamber **Bf** and a rear air chamber **Br** similar to those of the first embodiment (FIG. 2) are formed. The front air chamber **Br** is provided by a space enclosed by the inner surface of the front wall of the housing **A2** and the front surface side (diaphragm **23** side) of the speaker **SP**. The rear air chamber **Br** is provided by a space enclosed by the inner surface of the rear and side walls of the housing **A2** and the rear surface side (yoke **20** side) of the speaker **SP**. A plurality of sound holes **12** are formed in the front wall of the housing **A2**.

As shown in FIGS. 14A through 14C, the acoustic tube **40**, which has a rectangular shape in cross section, is formed along the inner walls of the rear air chamber **Br** at the back of the speaker **SP** over substantially three-quarter of the circumference of the rear air chamber **Br**. The acoustic tube **40** has an opening at one end **40a** and a closed end at the other end **40b**. The overall length **L** of the acoustic tube **40** is set to an odd multiple of one quarter of a wavelength, which is determined by the sonic velocity and a frequency **f** of which sound pressure level is desired to be increased. In this embodiment, the acoustic tube **40** has a length of one quarter of the wavelength determined by the frequency where an increase in sound pressure level of the audio output of the speaker **SP** is needed and the sonic velocity. In detail, since an open-end correction is performed on the acoustic tube **40**, where sonic velocity is set to C

and an open-end correction value is set to σ (=0.8**d**, wherein **d** is an open-end diameter of the acoustic tube **40**), the overall length is determined from an equation described below:

$$f = [C / \{4 (L+\sigma)\}]$$

In this embodiment, the overall length L of the acoustic tube **40** is set 95.2 mm and a cross-sectional area of the acoustic tube **40** is set to 4.2 mm^2 (equivalent to $\phi = 2.3$). By forming the acoustic tube **40** along the inner walls of the rear air chamber **Br**, it is possible to minimize a volume reduction of the rear air chamber **Br** caused by the formation of the acoustic tube **40**. In addition, with the acoustic tube **40** thus provided, the sound pressure level of the speaker **SP** can be increased in a desired frequency band (at around 700 Hz in this embodiment). As a result, it is possible to improve the sound quality and the electro-acoustic conversion efficiency of the speaker **SP**.

By utilizing the fact that input impedance becomes extremely small at the resonance frequency (i.e., a frequency at which the overall length of the tube is equal to an odd-number multiple of an approximately quarter (1/4) wavelength) of a closed tube, the acoustic tube **40** can reduce the sound leakage from the rear air chamber **Br** to the outside of the housing. That is, the sound waves radiated from the rear surface of the speaker **SP** is reflected toward the rear surface side of the speaker **SP**.

Referring to FIG. 15 through FIG. 18, effects of the acoustic tube **40** provided in the housing **A2** will be explained below in comparison with the intercom device **A** of the first embodiment, which has no acoustic tube in the housing **A1**, and an intercom device having an infinite baffle, which will be explained in detail below.

FIG. 15 shows frequency characteristics of the cancelled amount of the voice component from the speaker **SP** in the case of using the signal processing unit **10e** of intercom device **A** of the first embodiment, and FIG. 16 shows frequency characteristics of the radiating sound pressure measured in front of the speaker **SP**. FIG. 15 and FIG. 16 also show results obtained in the case

where the housing has no acoustic tube as in the first embodiment, and the case where an ideal baffle plate is used as the housing.

FIG. 17 shows a baffle plate **C** having an infinite size as an example of the above-mentioned ideal baffle plate, but the baffle plate **C** is not realistic for mounting a real speaker thereon. Thus, in this embodiment, a standard baffle as specified in JIS C 5532 is used to measure a frequency equivalent to a lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C**. The lowest resonant frequency of the speaker **SP** mounted on the standard baffle is equivalent to that of the speaker **SP** that is supposed to be mounted on the baffle plate **C** (an ideal baffle plate). The cancelled amount (**Y1a** in FIG. 15) in the case where the speaker **SP** and the microphone board **MB1** are mounted on the standard baffle to form an intercom device is kept at 10 dB or more in a frequency band of 100 Hz to 10000 Hz, and the radiating sound pressure characteristics (**Y2a** in FIG. 16) in the same case indicate that the lowest resonance frequency **fo1** of the speaker **SP** is 600 Hz, which is an equally ideal characteristic as provided by the speaker **SP** alone. A standard closed box as specified in JIS C 5532 may be used to measure a frequency equivalent to the lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C** having an infinite size (an ideal baffle plate) as shown in Fig. 17.

In the case where a rear air chamber having no acoustic tube **40** is used, as in the rear air chamber **Br** of the first embodiment, the cancelled amount (**Y1b** in FIG. 15) of the voice component from the speaker **SP** is kept at 10 dB or more in the frequency band of 100 Hz to 10000 Hz. Thus, substantially the same cancelled amount as that of the case where the standard baffle is used can be obtained. However, in the case where the rear air chamber **Br** is formed as an enclosed space, radiating sound pressure characteristics comparable to those of the case where the standard baffle is used can be obtained when the volume of the rear air chamber **Br** is sufficiently larger. On the other hand, when the volume of the rear air chamber **Br** is small, the mechanical equivalent stiffness

of the rear air chamber **Br** increases, and the lowest resonance frequency of the speaker **SP** becomes higher. As higher the lowest resonance frequency, the radiating sound pressure reduces, so that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** deteriorate. For example, the radiating sound pressure characteristics (**Y2b** in FIG. 16) of the case where the rear air chamber **Br** of the first embodiment is used indicate that the lowest resonance frequency **fo2** of the speaker **SP** is 1200 Hz, which is noticeably higher than the lowest resonance frequency obtained when the standard baffle is used. Furthermore, the sound pressure level in a frequency band below 800 Hz is lower by about 5 to 20 dB than that of the case where the standard baffle is used. It means that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are deteriorated in this frequency band.

The radiating sound pressure characteristics (**Y2c** in FIG. 16) of the case where the volume of the rear air chamber **Br** is made three times larger than that of the rear air chamber **Br** of the first embodiment indicate that the lowest resonance frequency **fo3** of the speaker **SP** is 800 Hz, and the sound quality of the speaker **SP** is improved as compared with the case of using the housing **A1** of the first embodiment. However, the overall size of the housing increases with the increase in volume of the rear air chamber **Br**, which makes it difficult to downsize the intercom device. Thus, in this embodiment, by providing the acoustic tube **40** in the rear air chamber **Br** having a small volume, the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are improved, while making the intercom device smaller.

The overall length **L** of the acoustic tube **40** is set so that the resonance frequency of the acoustic tube **40** is located between the lowest resonance frequency **fo2** (=1200 Hz) of the case where the rear air chamber **Br** having no acoustic tube **40** is used, and the lowest resonance frequency **fo1** (=600 Hz) of the case where the standard baffle is used. By setting the overall length **L** of the acoustic tube **40**, as described above, the lowest resonance frequency **fo5** of

the speaker **SP** in the case where the housing **A2** having the acoustic tube **40** is used is set to about 700 Hz between **fo1** and **fo2**.

FIG. 18 shows frequency characteristics of the radiating sound pressure of the intercom device **A'**, which is provided with the housing **A2** and the sound tube **40** of the present embodiment, and the radiating sound pressure of the intercom device **A** of the first embodiment, which has no acoustic tube. The characteristics **Y3a** of the intercom device **A'** in FIG. 18 indicate that the sound pressure level in a frequency band not larger than 800 Hz is increased by about 5 to 10 dB, as compared with the characteristics **Y3b** of the intercom device **A**, and the lowest resonance frequency is reduced from **fo4** (=1200 Hz) to **fo5** (= about 700 Hz). That is, improvements in the speaker efficiency and the sound quality can be achieved. For example, the sound pressure level at around 700 Hz is increased by about 10 dB, and the improvement of the speaker efficiency brought by the increase of the sound pressure level corresponds to a 70% reduction in input current and a 90% reduction in input power of the speaker **SP**. Thus, the sound pressure level of the speaker **SP** increases around the lowest resonance frequency of the speaker **SP**. Therefore, by setting the overall length of the acoustic tube **40** to an appropriate length, and shifting the lowest resonance frequency of the speaker **SP** to a lower frequency side, it is possible to achieve improvements in sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As another preferred acoustic tube of the present invention, an acoustic tube **41** that communicates with the rear air chamber **Br** may be provided at the outside of a housing **A3**. For example, the acoustic tube **41** can be obtained by attaching a tubular member to into a hole (not shown) formed in the housing **A3**, as schematically shown in FIG. 19. When tubular member is made of urethane or the like, the overall length **L** of the acoustic tube **41** can be easily changed, if desired.

As a further modification of this embodiment, two or more acoustic tubes may be formed in the rear air chamber **Br**. As an example, three acoustic tubes

42, **43**, **44** can be formed in the rear air chamber **Br** of a housing **A4**, as schematically shown in FIG. 20. In this example, the acoustic tubes (**42**, **43**, **44**) have different lengths from each other. This means that they have mutually different resonance frequencies. Accordingly, the sound pressure level of the speaker **SP** is increased at a plurality of frequencies. By setting the overall lengths of the respective acoustic tubes (**42**, **43**, **44**) to appropriate values, it is possible to provide desired sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As shown by a dotted line in FIG. 21, a sound absorbing material **45** such as a nonwoven fabric may be provided at or in the vicinity of the open end **40a** of the acoustic tube **40**. By use of the sound absorbing material **45**, it is possible to provide an enhanced sound absorbing effect, and perform a fine adjustment of the resonance frequency of the acoustic tube **40**.

(Third Embodiment)

The present embodiment provides a wiring system using an intercom device **101** of the present invention. That is, this wiring system enables power and signal transmissions between electric devices spaced away from each other in a building structure.

As shown in FIG. 22, the wiring system comprises a power line **Lp** and an information line **Ls** installed in the building structure, which are connected to a commercial power source **AC** and the Internet network **NT** through a distribution board, switch boxes **102** embedded in wall surfaces of the building structure, base units **103** mounted on the switch boxes, and connected to the power line **Lp** and the information line **Ls**, and function units **104**, each of which is detachably connected to a desired one of the base units **103**, and has at least one of functions for supplying electric power from the power line **Lp**, outputting information from the information line **Ls**, and inputting information into the information line **Ls** in a connected state with the desired base unit. An intercom device **101** of the present invention, which is preferably provided by

the intercom device (**A**, **A'**) of the first or second embodiment described above, can be regarded as one of the function units **104**. In the present description, the wall surfaces of the building structure are not limited to surfaces of sidewalls standing between adjacent rooms. That is, the meaning of the wall surfaces includes outdoor and indoor wall surfaces, and the indoor wall surfaces comprise ceiling and floor surfaces as well as the sidewall surfaces. In the drawings, **MB** designates a main circuit breaker, **BB** designates a branch circuit breaker, and **GW** designates a gateway (e.g., a router or a built-in hub).

As shown in FIGS. 23 and 24, each of the base units **103** is formed with a gate housing **130** having terminals (**131a**, **132a**, **131b**, **132b**) connected to the power supply line **Lp** and the information line **Ls**, and a main housing **135** detachably connected to the function unit **104**. Theses housings can be made of a synthetic resin having electrical insulating characteristics (e.g., a non-crystalline plastic such as ABS resin). The terminals (**131b**, **132b**) are used to extend the wirings. The gate housing **130** and the main housing **135** have a pair of a module port **134** and a module connector **142**, which are detachably connected to each other to simultaneously establish both of supplying the electric power from the gate housing **130** to the main housing **135**, and making a signal transmission therebetween. In addition, when the function unit **104** has the module connector **142**, it can be connected to the gate housing **130** in place of the main housing **135**. If necessary, the gate housing **130** and main housing **135** may be integrally formed.

Circuit components of the base unit **103** are designed to transmit the electric power and the information signal to the function unit **104**. For example, the base unit **103** is provided with an AC/AC converter **160**, DC power section **161**, transceiver section **162**, E/O converter **163**, O/E converter **165**, and a function portion **167**.

The AC/AC converter **160** converts commercial AC voltage to a lower AC voltage having an increased frequency, and applies the lower AC voltage to a coil

172. The DC power section **161** generates an operating voltage of the internal circuit components from a stable DC voltage obtained by rectifying and smoothing the lower AC voltage. The transceiver section **162** transmits and receives the information signal for enabling the mutual communication through the information line **Ls**. The E/O converter **163** converts the information signal received from the information line **Ls** to an optical signal, and outputs the optical signal though a light emitting device (LED) **164**. On the other hand, the O/E converter **165** receives the optical signal provided from the outside, e.g., the function unit **104** by a light receiving device (PD) **166**, and converting the received optical signal into the information signal to transmit it to the transceiver section **162**. The base unit **103** shown in FIG. 23 has an outlet tap as the function portion **167**. Alternatively, a sensor device or a controller may be formed as the function portion **167**.

The module port **134** formed at a front surface of the gate housing **130** is composed of an electric power port **134a** for supplying the electric power and an information signal port **134b** for accessing the information line **Ls**, as shown in FIG. 25B. Arrangement and shapes of the electric power port **134a** and the information signal port **134b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25B, each of the electric power port **134a** and the information signal port **134b** is configured in a substantially rectangular shape such that they are arranged in parallel to each other.

On the other hand, the module connector **142** formed at a rear surface of the main housing **135** is composed of an electric power connector **142a** and an information signal connector **142b**, as shown in FIG. 25A. Arrangement and shapes of the electric power connector **142a** and the information signal connector **142b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25A, each of the electric power connector **142a** and the information signal connector **142b** is configured

in a substantially rectangular shape such that they are arranged in parallel to each other.

In this embodiment, the module port **134** has a guide portion **133** such as a ring-like wall or a ring-like groove extending around the electric power port **134a** and the information signal port **134b**. This guide portion **133** is formed to be engageable to an engaging portion **145** such as a ring-like wall of the main housing **135**, which is formed around the electric power connector **142a** and the information signal connector **142b**. By simply engaging the engaging portion **145** to the guide portion **133**, the electric power connector **142a** and the information signal connector **142b** can be simultaneously connected to the electric power port **134a** and the information signal port **134b**. Therefore, there advantages that the main housing **135** can be easily attached to or removed from the gate housing **130** with improved connection reliability. A module connector having the same structure described above may be formed in the function unit **104**. The module port **134** and the module connector **142** may be formed by female and male connectors.

In the present embodiment, the gate housing **130** is directly fixed to the switch box **102**. If necessary, the gate housing **130** may be fixed to the switch box **102** through an attachment member **175**, as shown in FIG. 26. For example, this attachment member **175** is formed by a rectangular frame **176** with a window hole **177**. In this case, the gate housing **130** is mounted on the attachment member **175** by fitting a front portion of the gate housing into the window hole **177**, and then engaging hooks (not shown) formed at both sides of the gate housing to the rectangular frame **176**. The attachment member **175** mounting the gate housing **130** thereon is fixed to the switch box **102** by use of fixing screws. Alternatively, the gate housing **130** may be directly fixed to the wall surface by use of exclusive clamps without using the switch box **102**.

The function unit **104** is designed to provide a desired function by using the electric power supplied to the function unit **104** through the base unit **103** or

using the mutual communication of the information signal between the function unit **104** and the information line **Ls** through the base unit **103**. For example, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a high position near the ceiling, it preferably has a receptacle function of receiving a plug with hook of a lighting apparatus, security function such as a motion sensor, temperature sensor, and monitoring camera, or a sound function such as speaker. In addition, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a middle height, at which the function unit **104** can be easily operated by the user, it preferably has a switch function of turning on/off the lighting apparatus, control function for electric appliances such as air-conditioning equipments, a display function such as liquid crystal display or a timer function. The intercom device of the present invention is preferably connected as one of the function units **104** to the base unit **103** at the middle height. Moreover, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a low position near the floor, it preferably has a receptacle function for receiving a plug of an electric appliance such as electric vacuum cleaner, the sound function such as speaker, or a footlight function.

Specifically, as shown in FIG. 27, when a function section **181** of the function unit **104** is formed by a switch, operation data obtained by operating the switch is transmitted to a processing section **188** such as CPU through an I/O interface **189**. Then the processed data is sent to, for example, an infrared remote controller (not shown) through a transceiver section **187**, so that an electric appliance to be controlled is turned on/off by receiving a remote control signal emitted from the infrared remote controller. In addition, when the function section **181** is formed by a sensor, data detected by the sensor is transmitted as the information signal to the information line **Ls**, and then informed to the user by a required communicator.

The coil **172** wound around a core **170** of the base unit **103** shown in FIG.

23 is used as a power transmission means for supplying electric power to the function unit **104** in a noncontact manner. That is, the coil **172** of the base unit **103** provides an electromagnetic coupling portion that works as a first side of a transformer. On the other hand, as shown in FIG. 27, the function unit **104** has an electromagnetic coupling portion comprised of a coil **182** wound around a core **180**, which works as a second side of the transformer. Therefore, by forming electromagnetic coupling between the base unit **103** and the function unit **104**, a low AC voltage can be induced in the coil **182** of the function unit **104** to supply of electric power from the base unit **103** to the function unit **104**. Similarly, an electromagnetic coupling portion **X2** is formed at the opposite side of the electromagnetic coupling portion **X1**, and used to make a connection between the function units. In this embodiment, since the low AC voltage having the higher frequency than the commercial AC voltage is obtained by the AC/AC converter **160**, there is an advantage that the electromagnetic coupling portions used as the transformer can be downsized.

In addition, the light emitting device (LED) **164** of the base unit **103** shown in FIG. 23 is used to transmit an optical signal as the information signal to the function unit **104** in a noncontact manner. On the other hand, as shown in FIG. 27, a light receiving device (PD) **186** is disposed in the function unit **104** such that the light emitting device **164** is in a face-to-face relation with the light receiving device **186** when the function unit **104** is connected to the base unit **103**. Similarly, to transmit the optical signal as the information signal from the function unit **104** to the base unit **103**, the function unit **104** has a light emitting device (LED) **184**, which is disposed in the face-to-face relation with the light receiving element (PD) **166** when the function unit **104** is connected to the base unit **103**. Thus, each of the base unit **103** and the function unit **104** has the pair of the E/O converter (**163**, **183**) and the O/E converter (**165**, **185**) as an optical coupling portion to enable the mutual communication of the information signal therebetween in a noncontact manner.

As shown in FIG. 23, the electromagnetic coupling portion **X** used to supply the electric power and the optical coupling portion **Y** used for the mutual communication of the information signal are disposed at a side surface of the base unit **103** to be spaced from each other by a constant distance. Arrangements and shapes of the electromagnetic coupling portion **X** and the optical coupling portion **Y** are standardized (or stylized) to make the base unit **103** shareable among the function units **104**. In addition, the pair of the electromagnetic coupling portion **X** and the optical coupling portion **Y** may be formed at each of both sides of the function unit **104**, as shown in FIG. 27. That is, the optical coupling portion **Y1** formed at one side (e.g., left side) of the function unit **104** is composed of the light receiving device **186** located at the upper side and the light emitting device **184** located at the lower side, and the optical coupling portion **Y2** formed at the opposite side (e.g., right side) of the function unit **104** is composed of a light emitting device **194** located at the upper side and a light receiving device **196** located at the lower side. In this case, since the base unit **103** is connected to the one side of the function unit **104**, and another function unit or the intercom device of the present invention can be connected to the other side of the function unit **104**, there is an advantage of further improving function expandability of the wiring system.

The intercom device **101** of this embodiment can be connected to the base unit **103**, as shown in FIG. 28. In addition, internal components of the intercom device **101** of this embodiment are shown in FIG. 29. As understood from the comparison with the function unit **104** of FIG. 27, the intercom device **101** of FIG. 29 is substantially equivalent to a function unit having the components of the intercom device shown in FIG. 2 as the function section **181**.

Since this intercom device **101** has the stylized connectors as well as the function unit **104** and base unit **103** described above, it can be used in a desired combination with the base unit **103** and the function unit **104**. For example, as shown by the arrow ① in FIG. 30, the intercom device **101** can be

connected to the function unit **104** having the timer function, which is connected to the base unit **103**. Alternatively, as shown by the arrow ② in FIG 30, the intercom device **101** may be connected at its one side to the base unit **103**. At this time, the function unit **104** having the timer function can be connected to the other side of the intercom device **101**. Thus, there is a remarkable advantage that a layout of the function units **104** having various functions and the intercom device **101** can be changed with a high degree of freedom according to the user's needs.

In addition, when the intercom device **101** is operated in conjunction with at least one functions unit **104**, the wiring system of this embodiment can provide a higher order function in addition to the communication function. For example, when the intercom device **101** is connected to one of the base units **103**, and the function unit **104** having a sensor function (e.g., motion sensor or fire sensor) is connected to another one of the base units **103**, which is disposed at a proper location, the intercom device **101** can output an alarm sound by receiving alarm signal from the function unit. Therefore, the intercom device **101** of this case has an alarm generating function for disaster or crime prevention as well as the communication function. In addition, since it is not needed to individually prepare a speaker for generating the alarm sound, there is a further advantage of achieving an improvement in cost performance of the wiring system as a whole.

Next, means for mechanically coupling between the base unit **103** and the function unit **104** (or the intercom device **101**) is explained.

For example, the base unit **103** shown in FIG. 31 has a cosmetic cover **112**, which is detachably attached to a front surface of the gate housing **135**, and horizontal guide rails **114** formed at upper and lower portions of the gate housing. The numeral **115** designates a stopper wall formed in a substantially center position in the longitudinal direction of the guide rail **114**. On the other hand, as shown in FIGS. 32A to 32C, the function unit **104** (or the intercom

device **101**) has concave portions **128** for accommodating joining members **150** therein, which are formed at both of its upper and lower sides, horizontal guide rails **124** extending in the concave portions **128**, and cover members **126** pivotally supported at upper and lower ends of the function unit **104**. In addition, as shown in FIG. 32D, the joining member **150** has a groove **152**, in which the horizontal guide rail (**114**, **124**) can be slidably fitted.

In this case, a cosmetic cover **112** is firstly removed from the gate housing **135** of the base unit **103**. Then, the electromagnetic coupling portion **X** and the optical coupling portion **Y** of the base unit **103** are connected to the electromagnetic coupling portion **X1** and the optical coupling portion **Y1** of the function unit **104** (or the intercom device **101**). Next, the cover member **126** is opened, and the joining member **150** is slid along the guide rail **124**, as shown by an arrow in FIG. 33. The slide movement of the joining member **150** is performed until the joining member **150** contacts the stopper wall **115**. As a result, the joining member **150** is engaged to the guide rail **114** of the base unit **103** over about half length of the joining member **150**. On the other hand, the remaining portion of the joining member **150** is still engaged to the guide rail **124** of the function unit **104** (or the intercom device **101**). Thus, by engaging the joining member **150** to both of the base unit **103** and the function unit **104** the base unit **103** can be mechanically coupled with the function unit **104** (or the intercom device **101**). Finally, the cosmetic cover **112** is attached to the base unit **103**, and the cover member **126** is closed.

According to this coupling means, since the joining member **150** is held between the cosmetic cover **112** and the gate housing **135**, it is possible to prevent accidental falling of the joining member **150**, and obtain the stable mechanical connection therebetween without spoiling the beauty of them. In addition, since the joining member **150** is always accommodated in the concave portion **128** of the function unit **104** (or the intercom device **101**), there is no worry about loss of the joining member **150**. This coupling means is also usual

in the case of mechanically coupling between the function units **104**, or between the function unit **104** and the intercom device **101**.

As an information-signal transmitting method available in the wiring system, a baseband transmission or a broadband transmission can be used. In addition, the protocol is not limited to a specific one. For example, sound and visual information signals may be transmitted and received according to JT-H232 packet to make the mutual communication between the intercom devices. In a control system, it is also preferred to use a routing protocol for a broadcast or a unicast that controlling can be performed at a control ratio of 1:1 or 1:N according to operation data. Alternatively, the protocol used between the base units may be different from the protocol used in the function unit or the intercom device connected to the base unit. In this case, a protocol conversion is preferably performed by the base unit.

In addition, the wiring system of the above embodiment is essential to use both of the information line and the power line installed in the building structure. As a modification of this embodiment, a power-line carrier communication may be used in the wiring system.

That is, the wiring system according to this modification is essential to install only the power line in the building structure. Therefore, the base unit is connected to only the power line. On the other hand, the base unit, the function unit and/or the intercom device need to have a transceiver unit for transmitting and receiving information signals in the power-line carrier communication manner. For example, when the transceiver unit is formed in the base unit, the electric power carrying the information signals thereon is separated to the electric power transmission and the information signal transmission by the base unit. Therefore, the same function unit and the intercom device used in the wiring system of the above embodiment are also available in this power-line carrier type wiring system.

Alternatively, the transceiver unit may be formed in the intercom device **101**, as shown in FIG. 34. In this case, the intercom device is detachably

connected to the base unit and/or the function unit by a power-line carrier type connector **Z**. In addition, the intercom device also comprises a PLC modem **210** for receiving the information signals carried by the power line, and transmitting information signals through the power line, a processing section **220** connected to the PLC modem, an I/O interface **230** between the processing section, and a function section **240** for providing the intercom function. Similar circuit components may be formed in the function unit and/or the base unit. As a modulation method for the power-line carrier communication, for example, it is possible to use a broad spectrum diffusion model, multi carrier model or an orthogonal Frequency division multiplexing (OFDM) model.

INDUSTRIAL APPLICABILITY

Thus, according to the present invention, the first microphone disposed in the front air chamber efficiently receives the voice output of the speaker, and the output of the first microphone is effectively used in the signal processing unit to remove the signal component corresponding to the voice output of the speaker from the output of the second microphone. Therefore, it is possible to improve the howling prevention effect. In addition, the intercom device can be downsized because a distance between the first microphone and the speaker becomes short. Furthermore, the intercom device is suitable for the wiring system with excellent function expandability, as described in the above embodiments.

Therefore, the present invention is expected to further increase the demand of the intercom device, and especially as one of the components of the wiring system of the next generation.

CLAIMS:

1. An intercom device comprising:
a housing;
a speaker accommodated in said housing such that a front air chamber is defined as a space enclosed by a front surface of said speaker and an inner surface of said housing, and a rear air chamber is defined as a space enclosed by a rear surface of said speaker and an inner surface of said housing;
a first microphone disposed such that its sound receiving surface faces the front air chamber;
a second microphone disposed such that its sound receiving surface faces outside of said housing; and
a signal processing unit configured to, when an audio output of said speaker is picked up by said second microphone, remove a signal component corresponding to the audio output of said speaker from an output signal of said second microphone by use of an output signal of said first microphone.

2. The intercom device according to claim 1, wherein the rear air chamber is the space hermetically-sealed by the rear surface of said speaker and the inner surface of said housing.

3. The intercom device according to claim 1, wherein said first microphone is arranged such that the sound receiving surface of said first microphone faces a diaphragm of said speaker.

4. The intercom device according to claim 1, wherein said second microphone is located adjacent to said speaker.

5. The intercom device according to claim 1, wherein said first microphone and said second microphone are mounted on a same circuit board.

6. The intercom device according to claim 5, wherein the circuit board is mounted on an outside surface of said housing such that the sound receiving surface of said first microphone faces the front air chamber and the sound receiving surface of said second microphone faces the outside of said housing.

7. The intercom device according to claim 1, further comprising an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

8. The intercom device according to claim 7, wherein said acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of said speaker is needed.

9. The intercom device according to claim 1, further comprising a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

10. The intercom device according to claim 7, wherein said acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant

frequency of said speaker that is supposed to be mounted on an infinite baffle and a lowest resonant frequency of said speaker mounted on the rear air chamber with the absence of said acoustic tube.

11. The intercom device according to claim 7, wherein said acoustic tube is formed along an inner surface of the rear air chamber.

12. The intercom device according to claim 7, further comprising a sound absorbing material disposed at the opening of said acoustic tube.

13. A wiring system using the intercom device as set forth in claim 1, comprising at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of said building structure, and connected to said transmission line;

wherein the intercom device has a first connector detachably connectable to a second connector formed in each of said base units,

when said first connector is connected to said second connector of one of said base units, the intercom device functions such that the information signals provided through said transmission line are output as audio information from said speaker unit, and an output of said signal processing unit is sent out through said transmission line.

14. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units, said function unit having a third connector detachably

connectable to one of said second connector of said base unit, and an extended connector formed in the intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, or when said first connector of the intercom device is connected to said second connector of said base unit, and said third connector of said function unit is connected to said extended connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided through said transmission line, outputting information signals provided through said transmission line, and sending out information signals through said transmission line.

15. The wiring system as set forth in claim 14, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

16. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units,

said function unit has a third connector detachably connectable to said second connector of said base unit and an extended connector detachably connectable to said first connector of said intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, and said extended connector of said function unit is connected to said first connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided from said transmission line through said base unit, outputting information signals provided from said transmission line through said base unit, and sending out

information signals to said transmission line through said base unit, and the intercom device operates such that the information signals provided from said transmission line through said base unit and said function unit are output as audio information from said speaker unit, and the output of said signal processing unit is sent out to said transmission line through said base unit and said function unit.

17. The wiring system as set forth in claim 16, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

18. The wiring system as set forth in claim 13, wherein said first and second connectors provide an electric power transmission between said base unit and the intercom device by means of electromagnetic coupling.

19. The wiring system as set forth in claim 13, wherein said first and second connectors provide a signal transmission between said base unit and the intercom device by means of optical coupling.

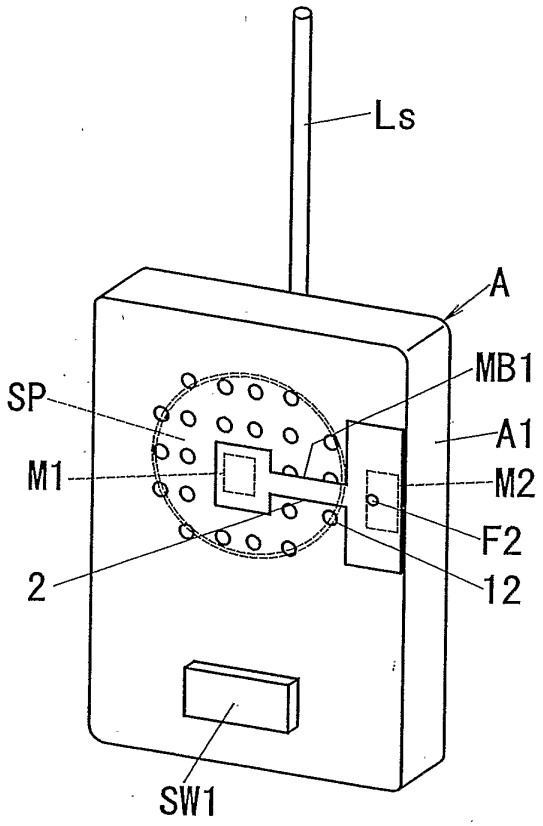


FIG. 1

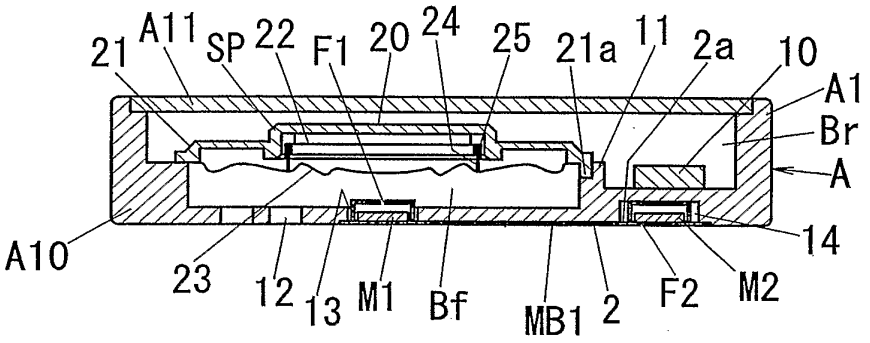

21
A11
SP
22
F1
20
24
25
21a
11
2a
10
A1
Br
A
14
A10
23
12
13
M1
Bf
MB1
2
F2
M2

FIG. 2

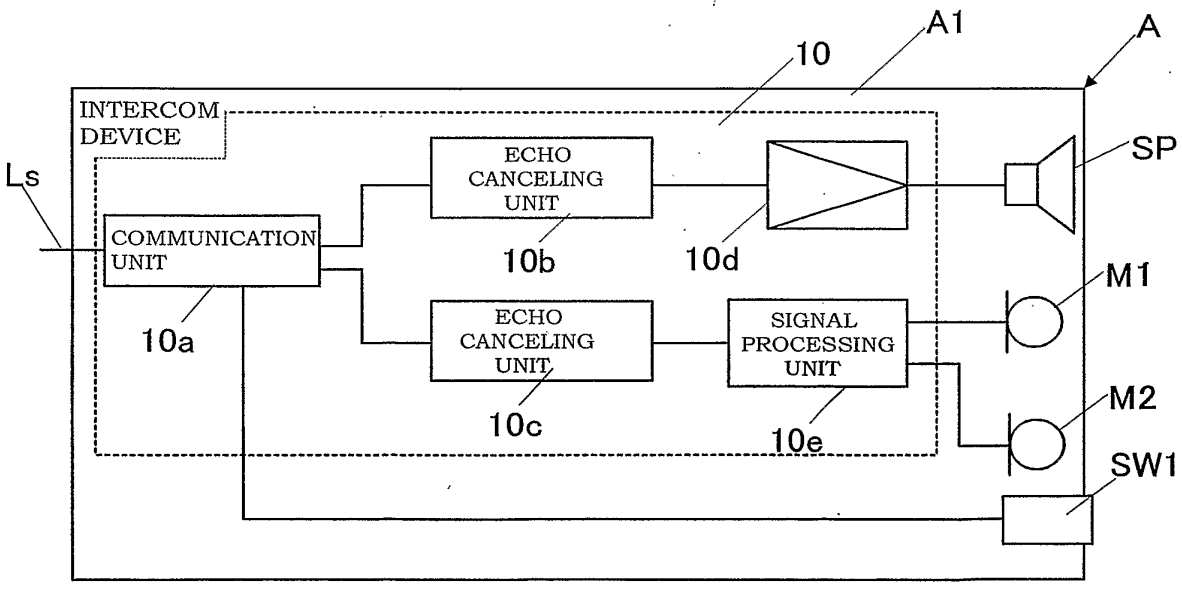

A1
10
A
INTERCOM DEVICE
Ls
ECHO CANCELING UNIT
10b
10d
SP
COMMUNICATION UNIT
10a
M1
ECHO CANCELING UNIT
10c
SIGNAL PROCESSING UNIT
10e
M2
SW1

FIG. 3

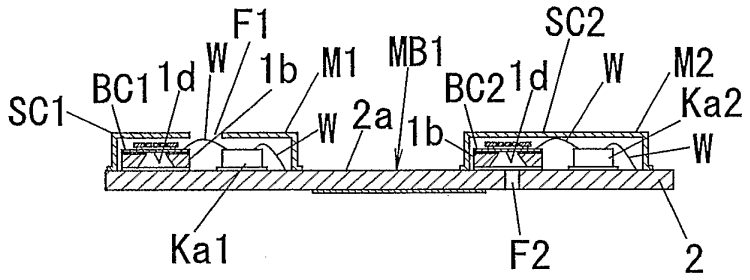

F1
W
1b
M1
MB1
SC2
W
M2
SC1
BC1
1d
W
2a
1b
BC2
1d
Ka2
W
Ka1
F2
2

FIG. 4

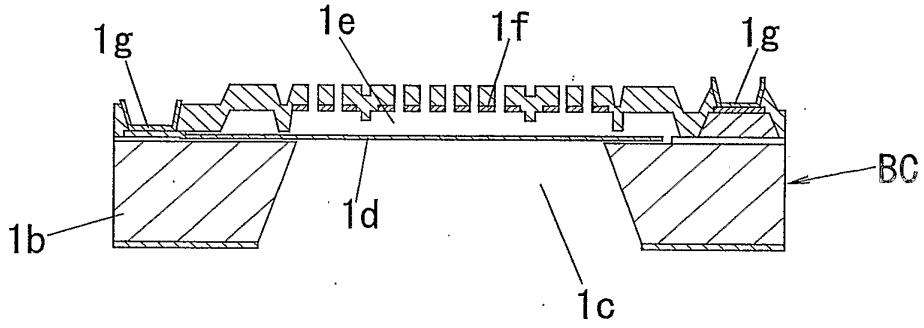

1g
1e
1f
1g
BC
1b
1d
1c

FIG. 5

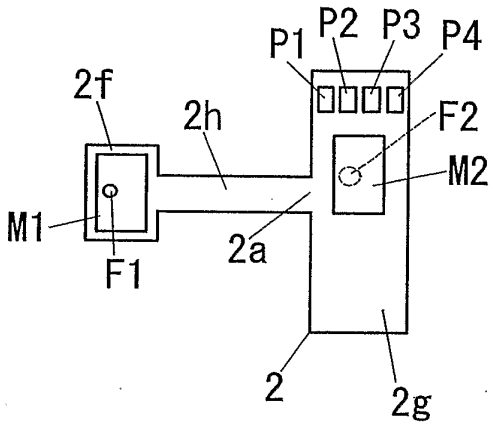


FIG. 6A

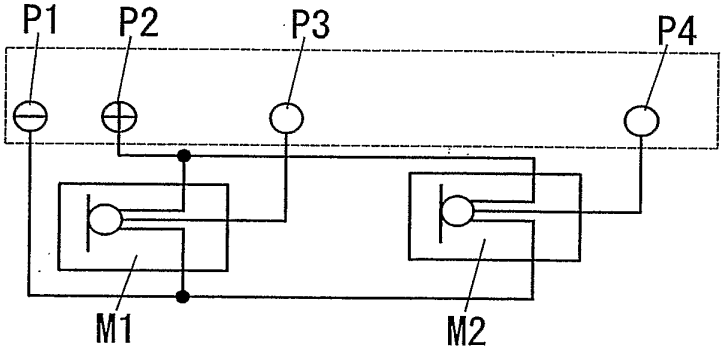


FIG. 6B

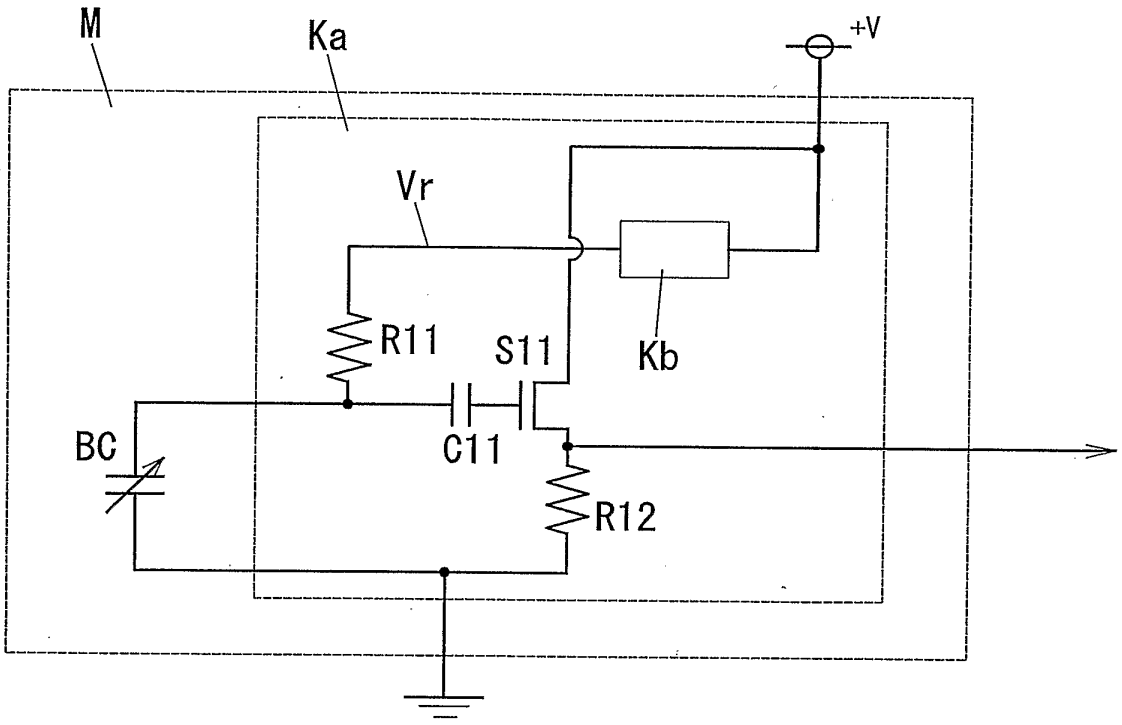


FIG. 7

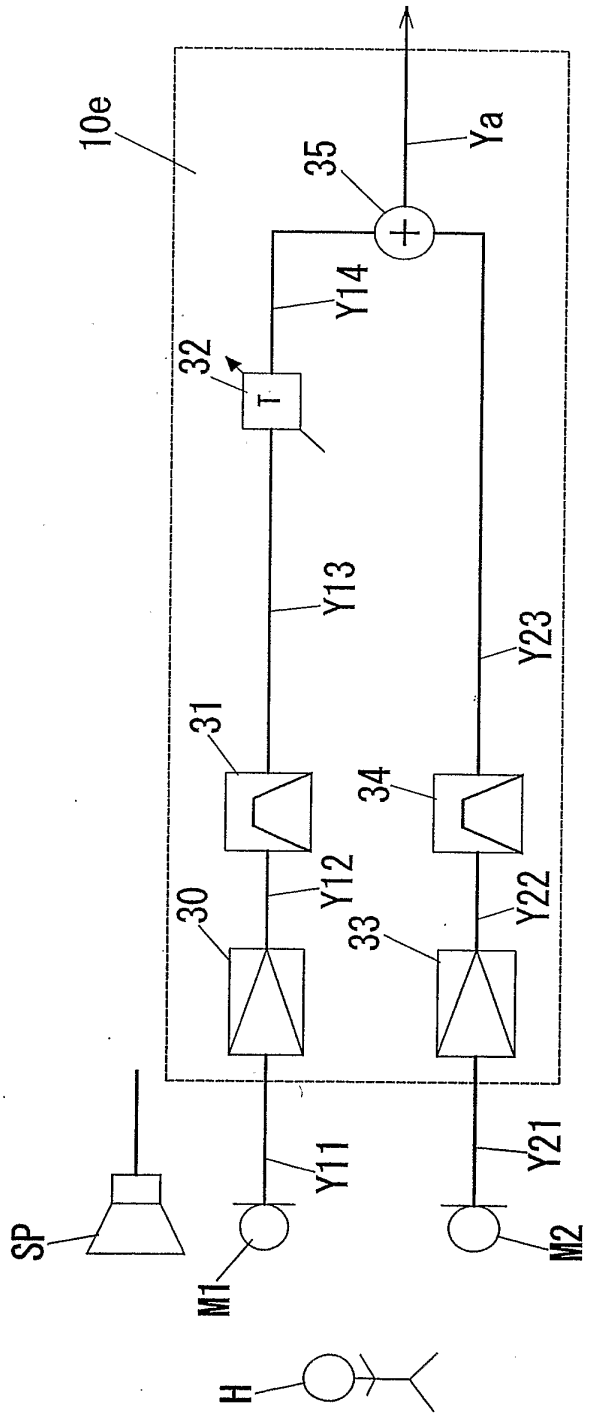


FIG. 8

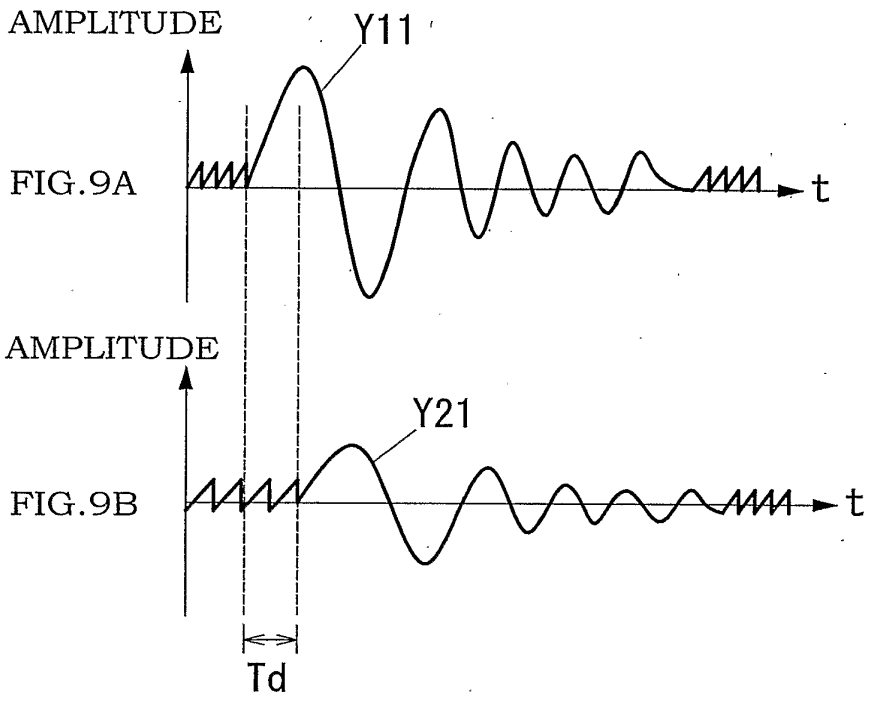


FIG.9A

FIG.9B

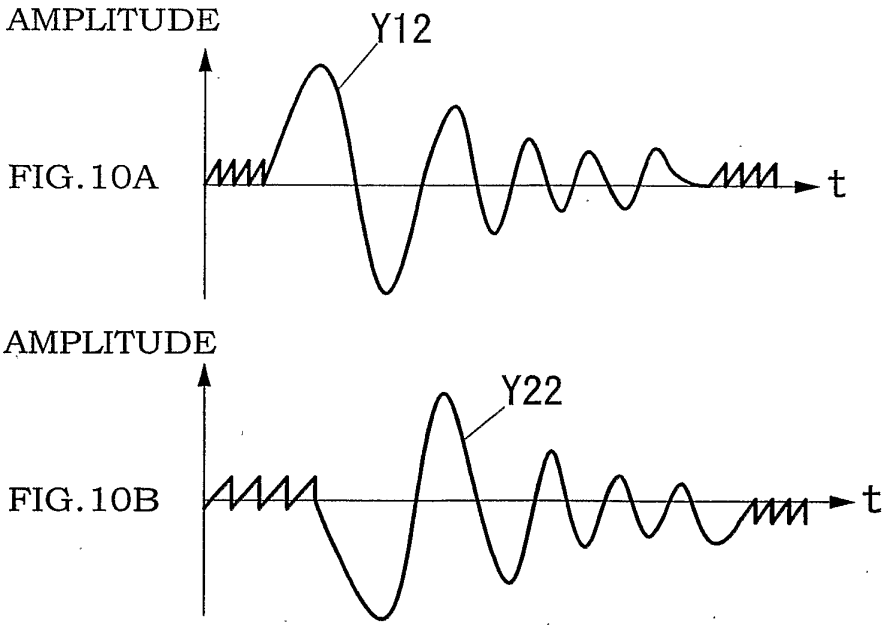


FIG.10A

FIG.10B

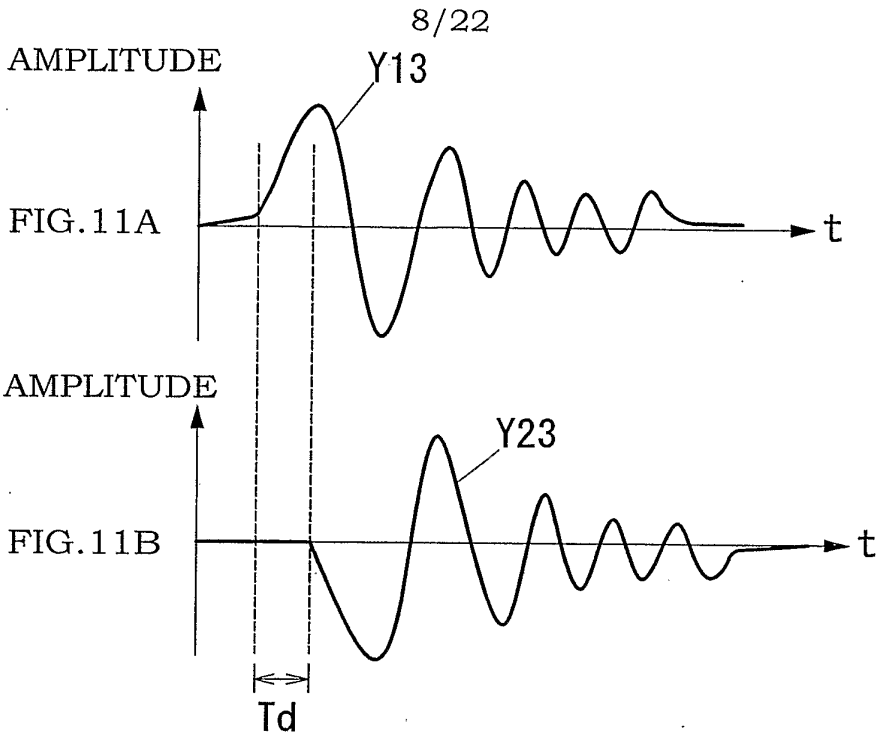


FIG.11A

FIG.11B

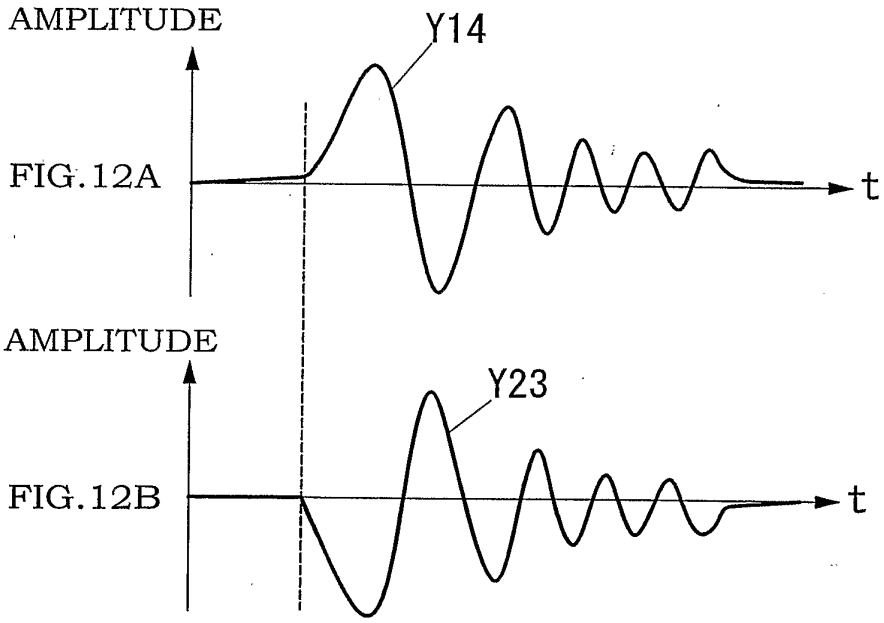


FIG.12A

FIG.12B

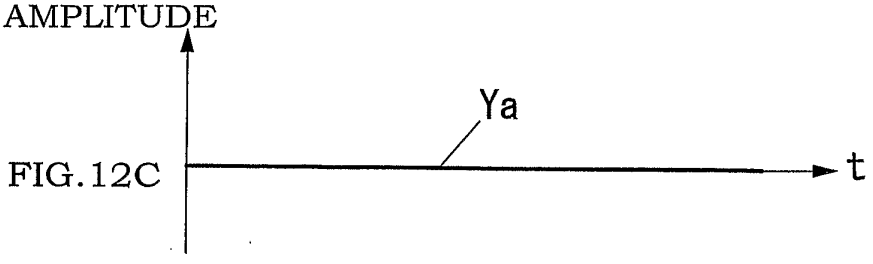
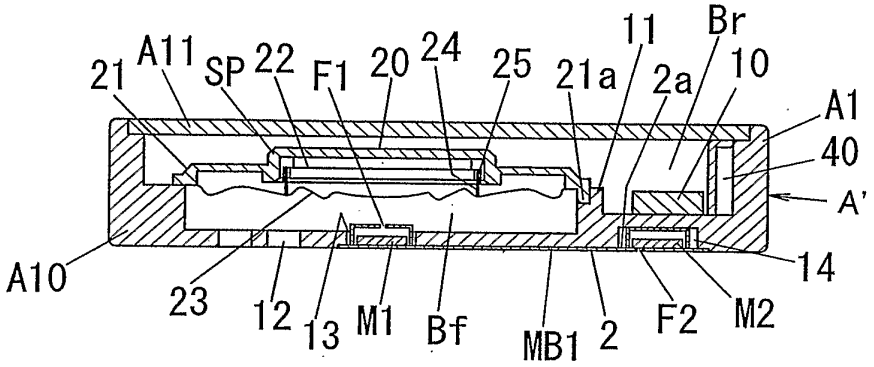


FIG.12C

FIG. 13



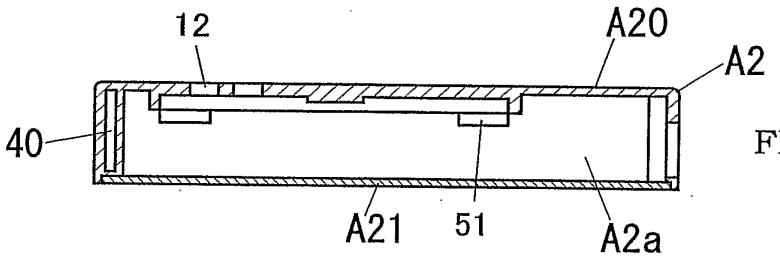
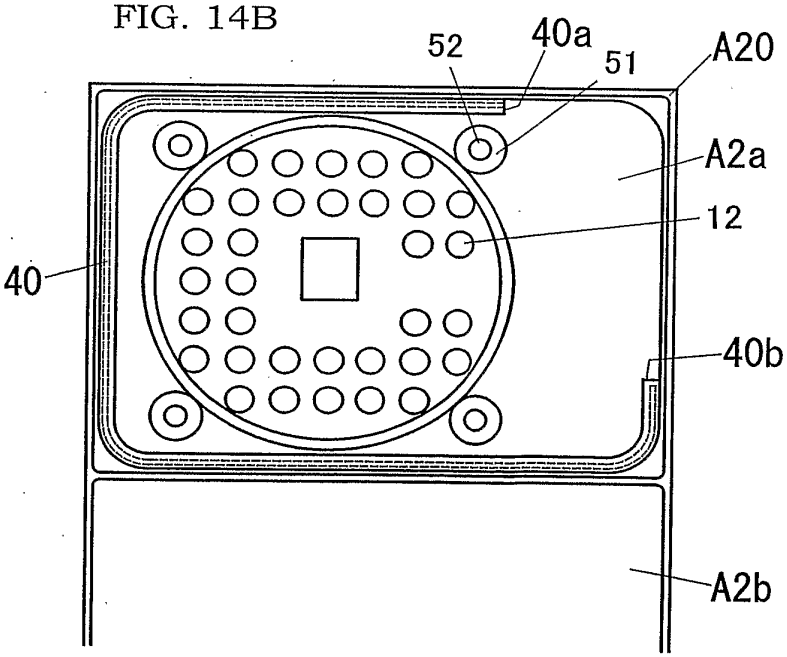


FIG. 14A

FIG. 14B



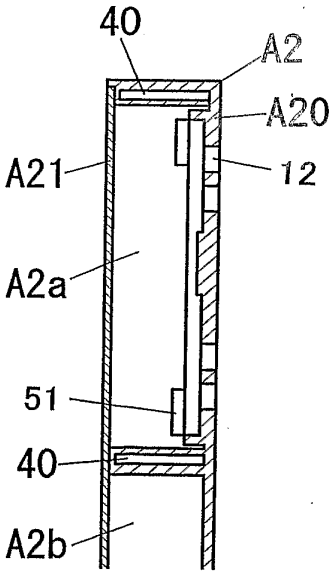


FIG. 14C

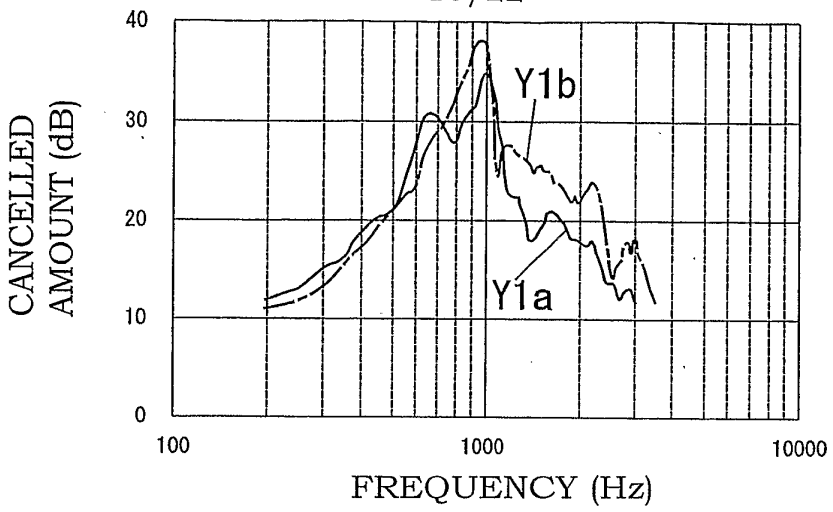


FIG. 15

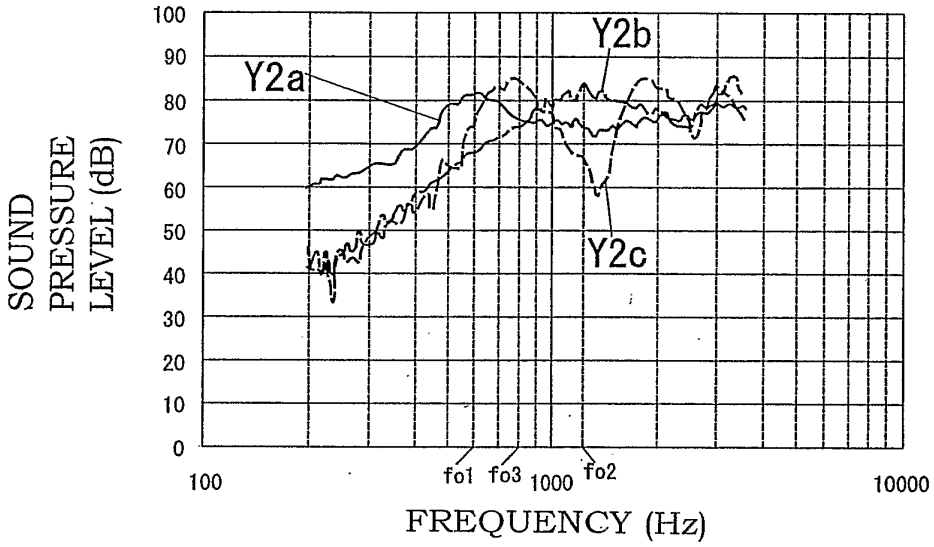


FIG. 16

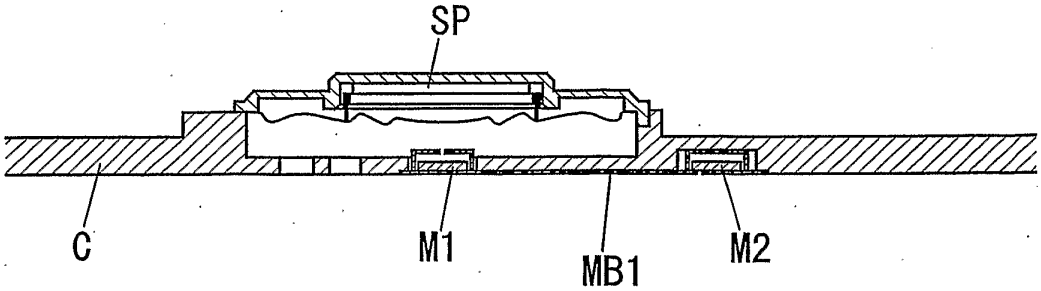


FIG. 17

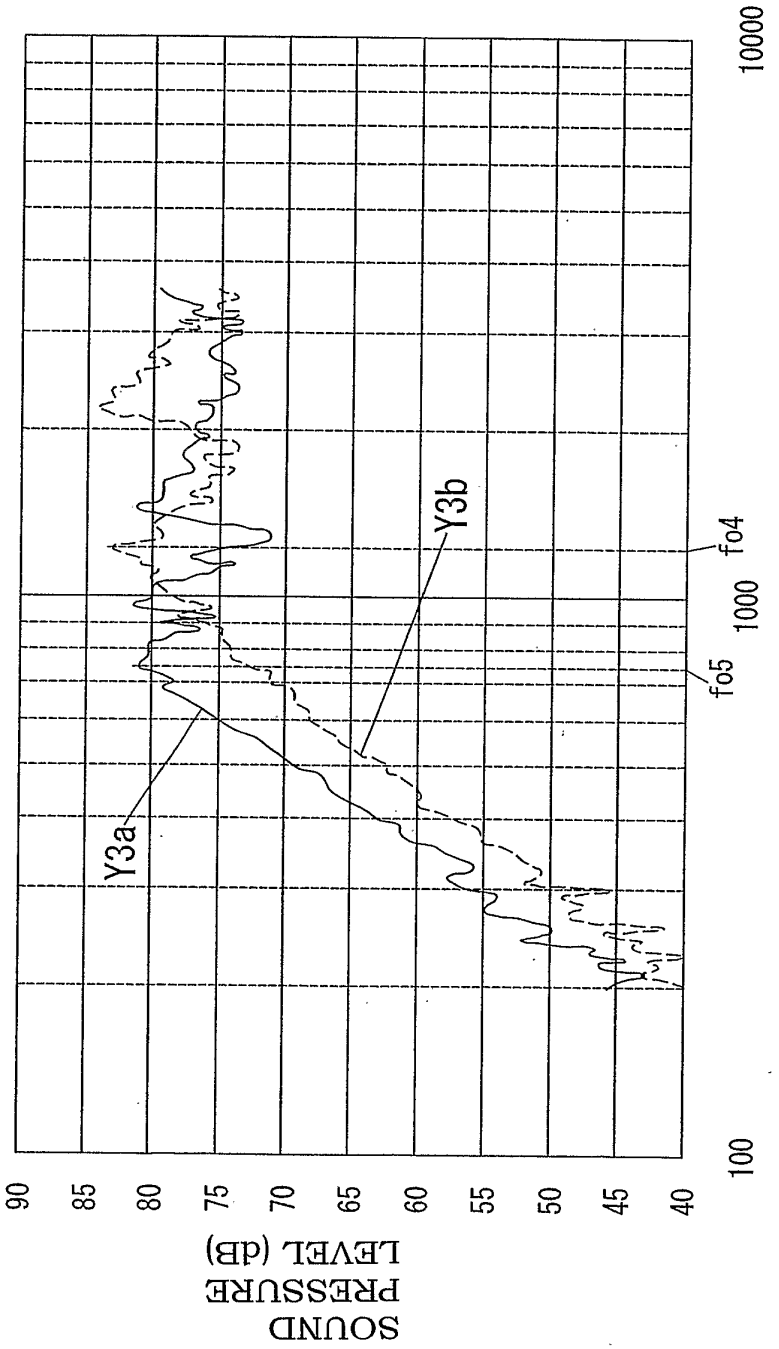

SOUND PRESSURE LEVEL (dB)
90
85
80
75
70
65
60
55
50
45
40
100
1000
10000
FREQUENCY (Hz)
fo4
fo5
Y3a
Y3b

FIG. 18

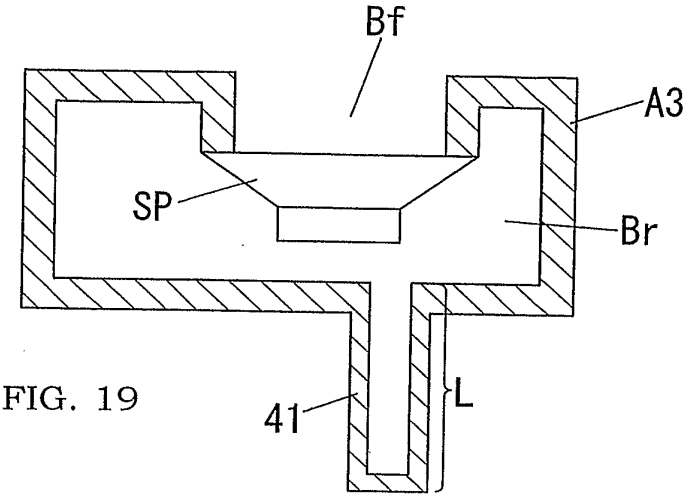


FIG. 19

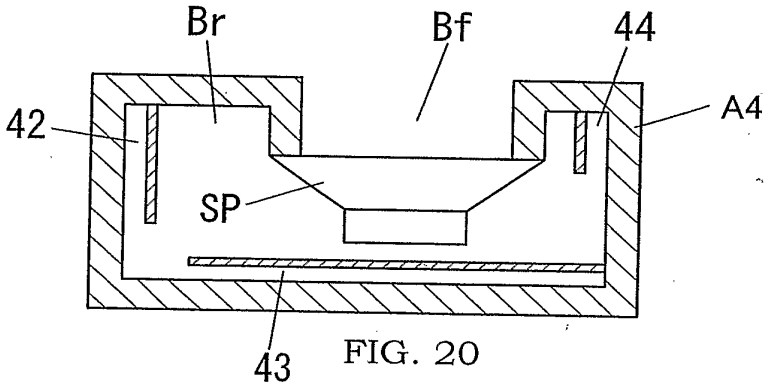


FIG. 20

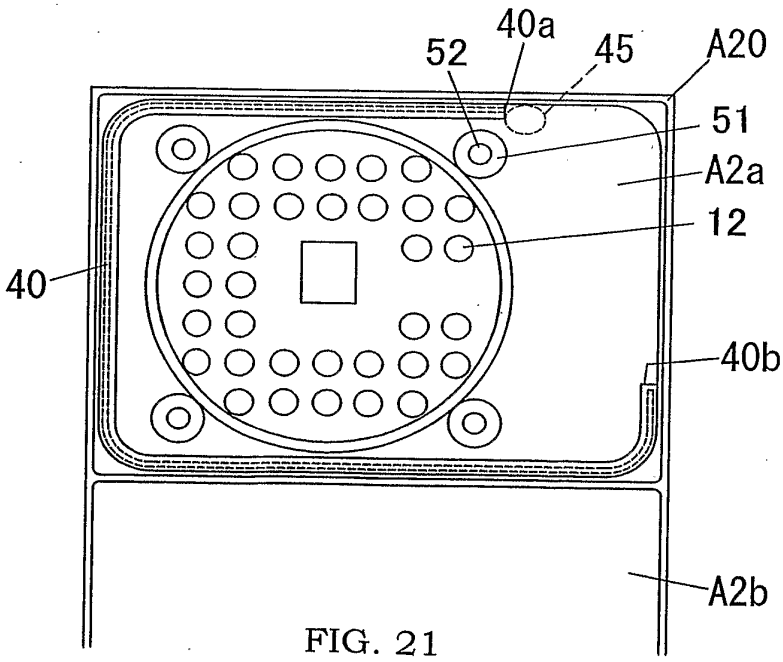


FIG. 21

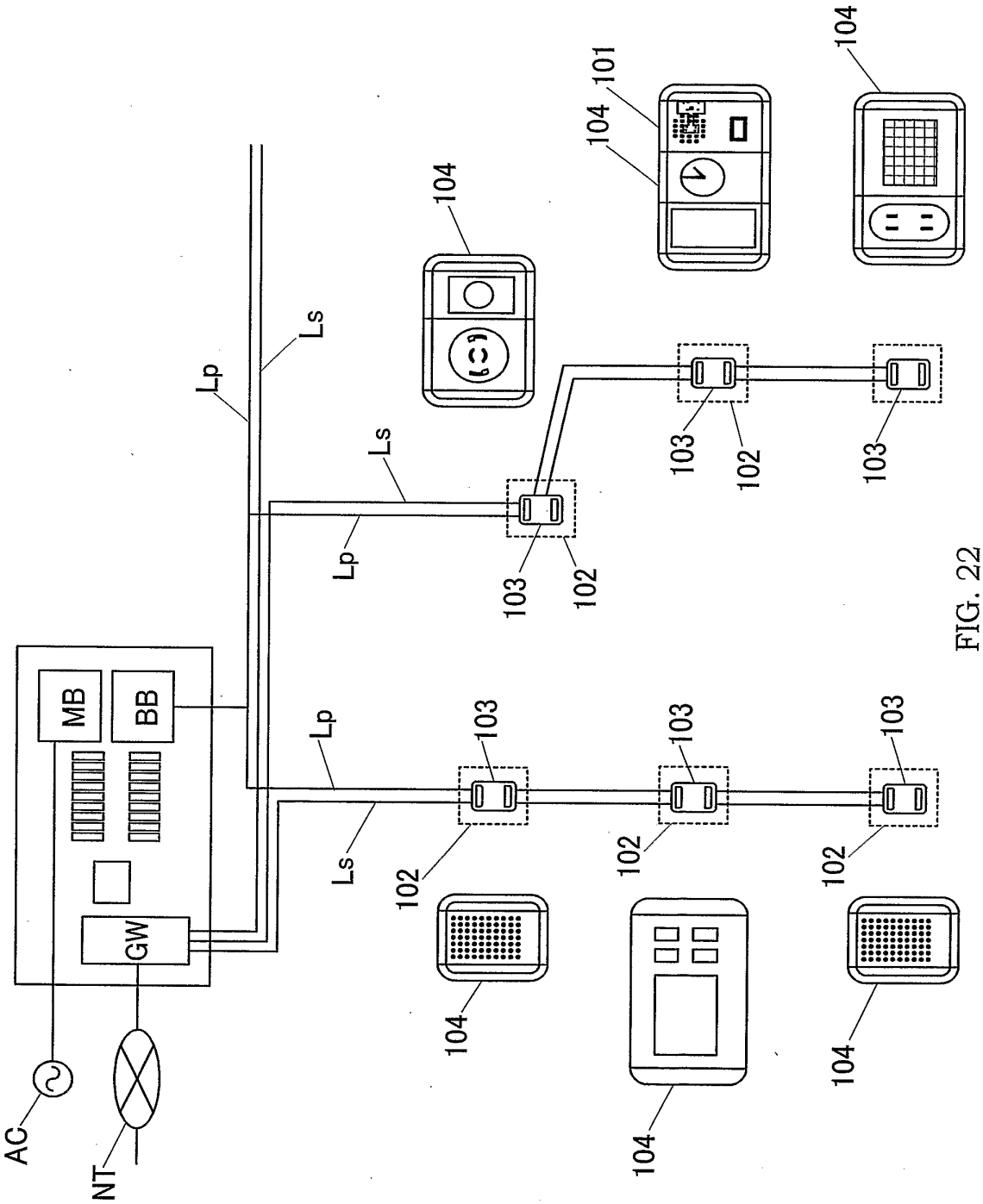


FIG. 22

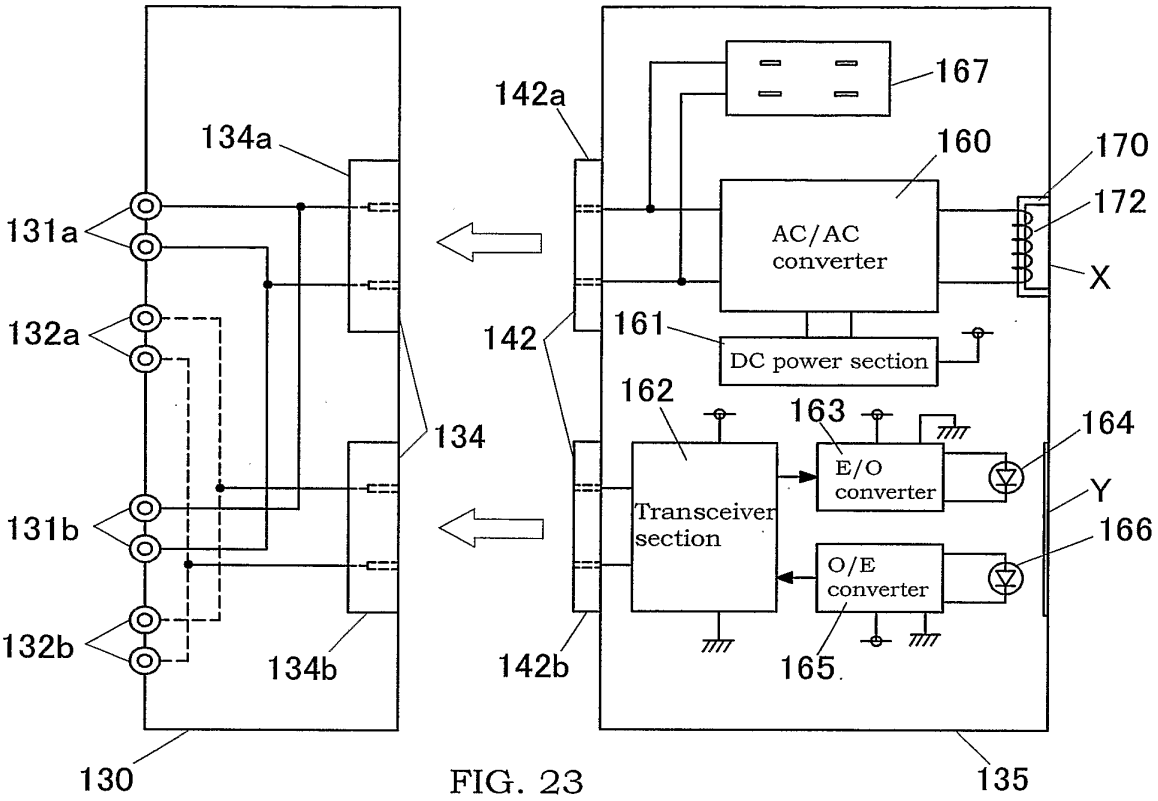

134a
131a
132a
131b
132b
134b
134
130
142a
167
160
170
172
AC/AC converter
X
142
161
DC power section
162
163
164
E/O converter
Y
Transceiver section
166
O/E converter
142b
165
135

FIG. 23

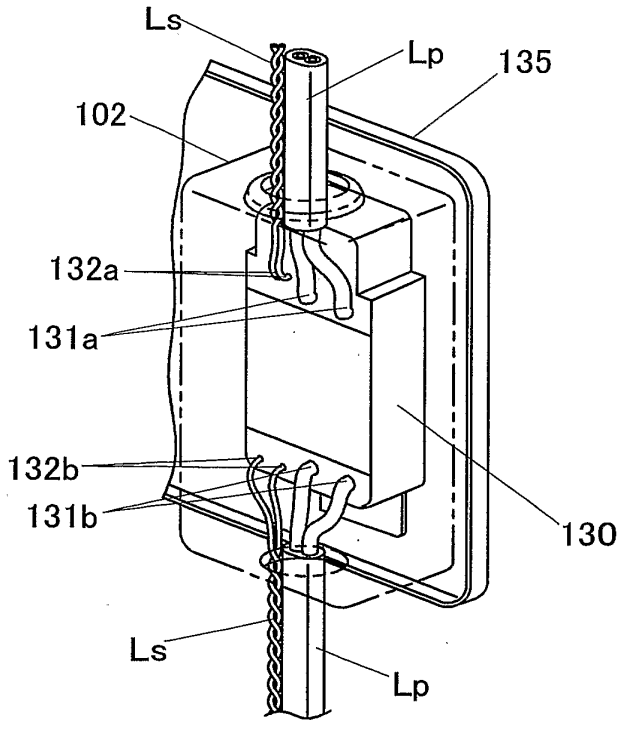

Ls
Lp
135
102
132a
131a
132b
131b
130
Ls
Lp

FIG. 24

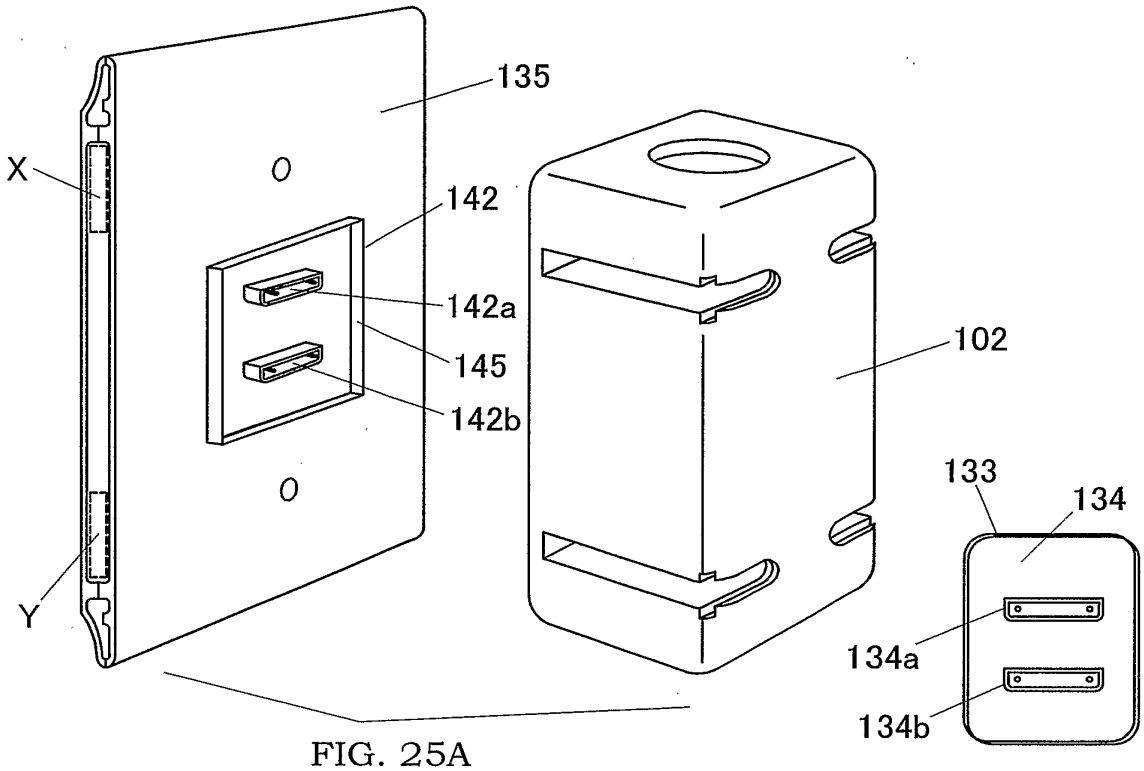


FIG. 25A

FIG. 25B

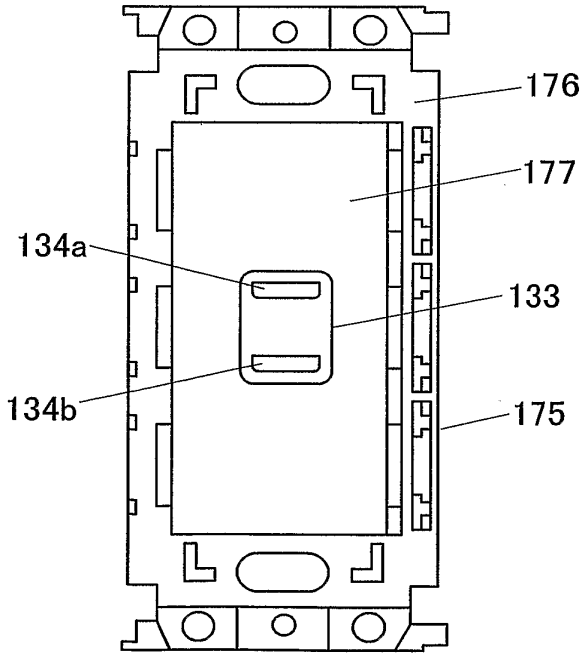


FIG. 26

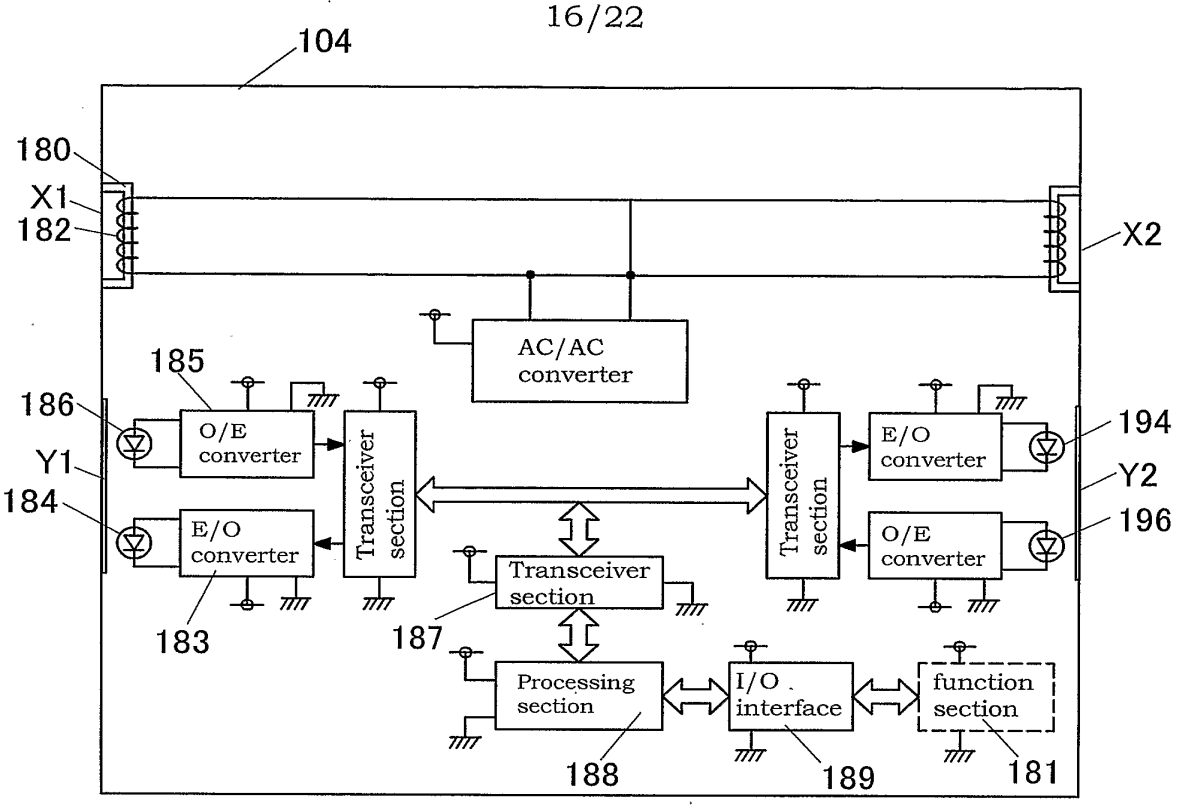

104
180
X1
182
X2
AC/AC converter
185
186
O/E converter
Transceiver section
Transceiver section
E/O converter
194
Y1
Y2
184
E/O converter
O/E converter
196
Transceiver section
183
187
Processing section
I/O interface
function section
188
189
181

FIG. 27

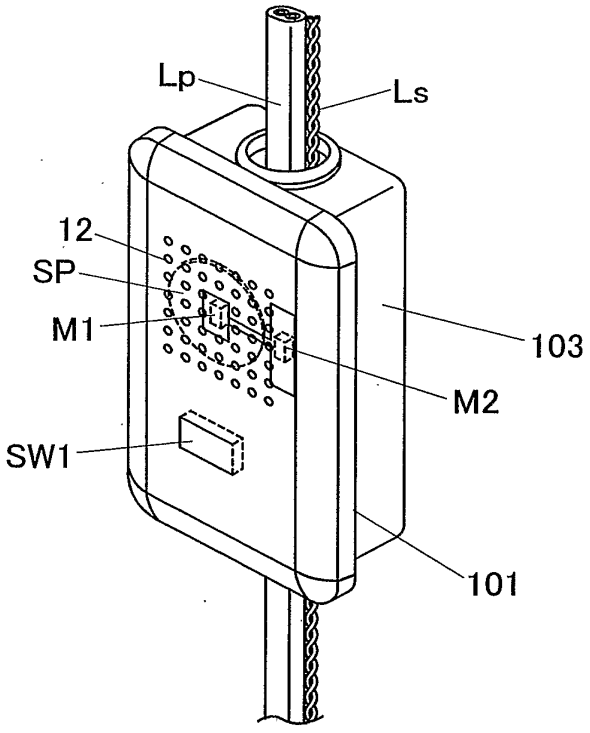

Lp
Ls
12
SP
M1
103
M2
SW1
101

FIG. 28

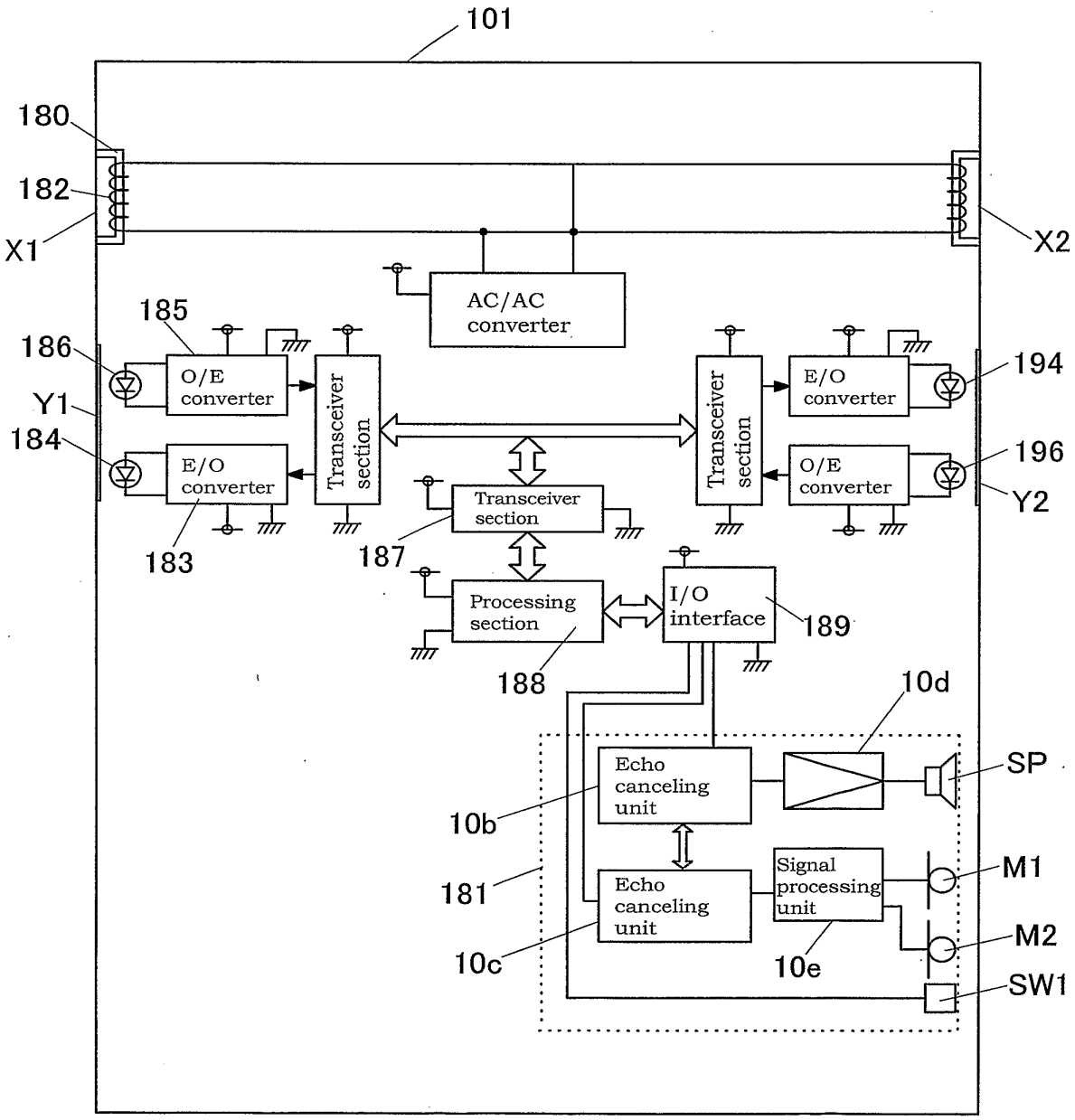

101
180
182
X1
X2
AC/AC converter
185
186
O/E converter
Transceiver section
Y1
184
E/O converter
183
187
Transceiver section
Transceiver section
E/O converter
194
O/E converter
196
Y2
Processing section
I/O interface
189
188
10d
Echo canceling unit
SP
10b
181
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1

FIG. 29

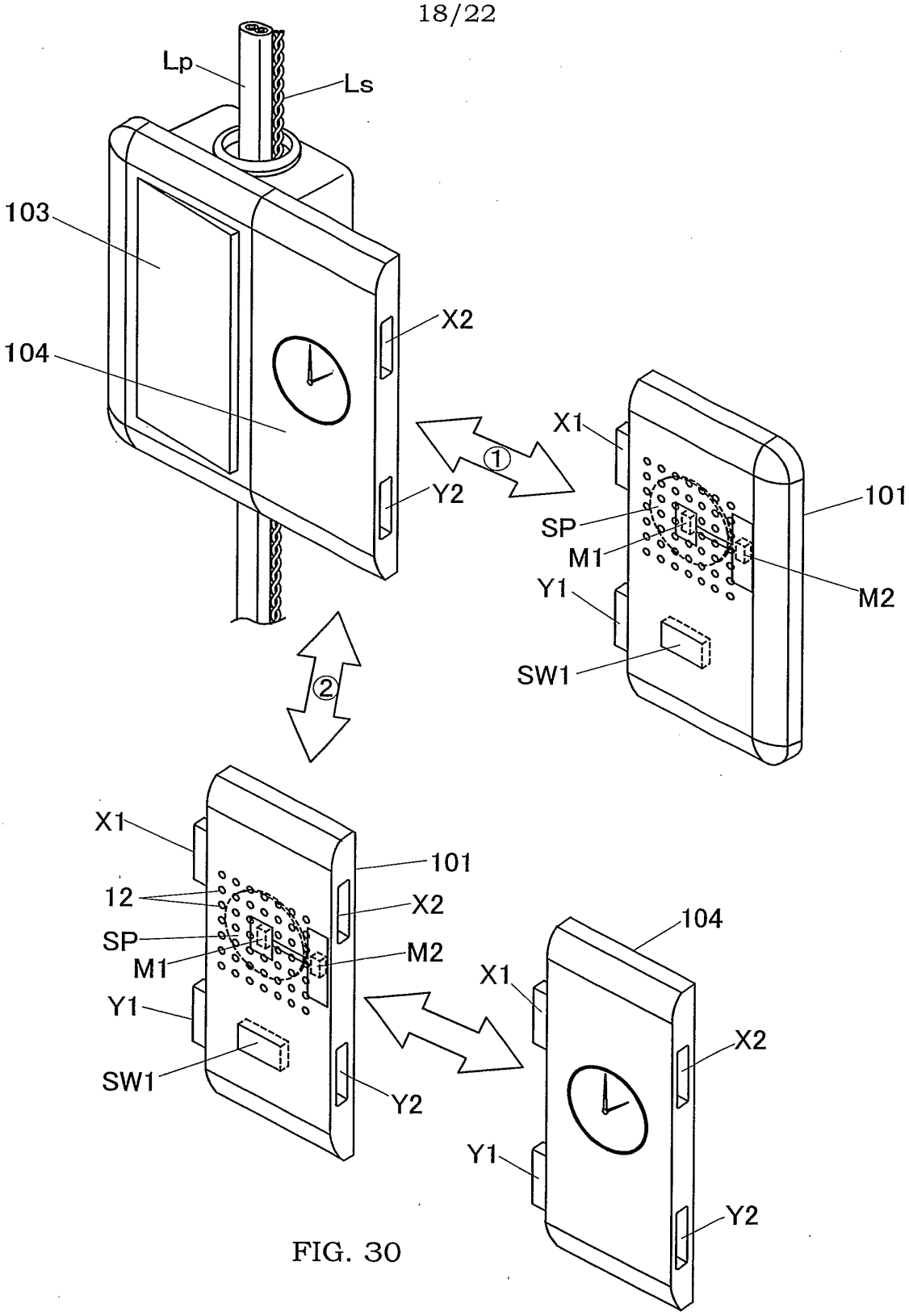

Lp
Ls
103
104
X2
Y2
①
X1
101
SP
M1
M2
Y1
SW1
②
X1
101
12
X2
SP
M1
M2
Y1
SW1
Y2
104
X1
X2
Y1
Y2

FIG. 30

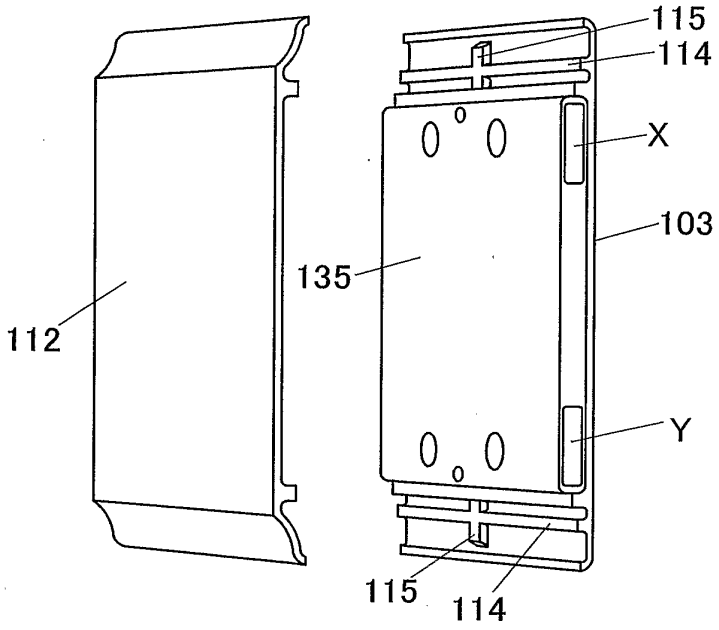


FIG. 31

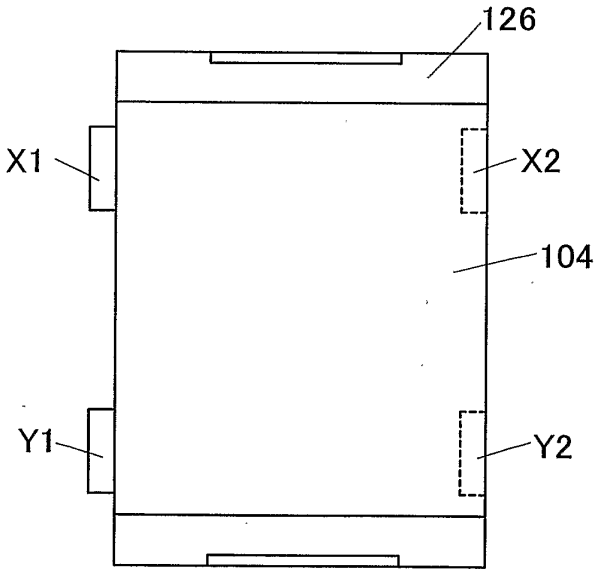


FIG. 32A

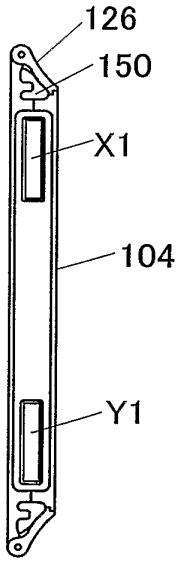


FIG. 32B

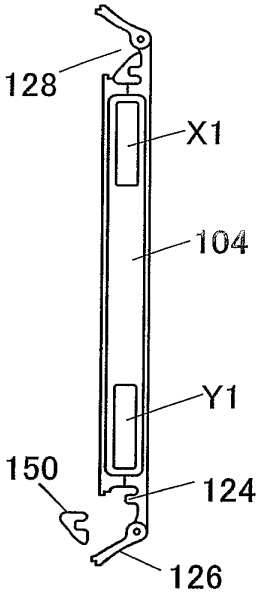


FIG. 32C

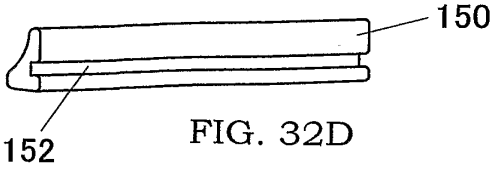


FIG. 32D

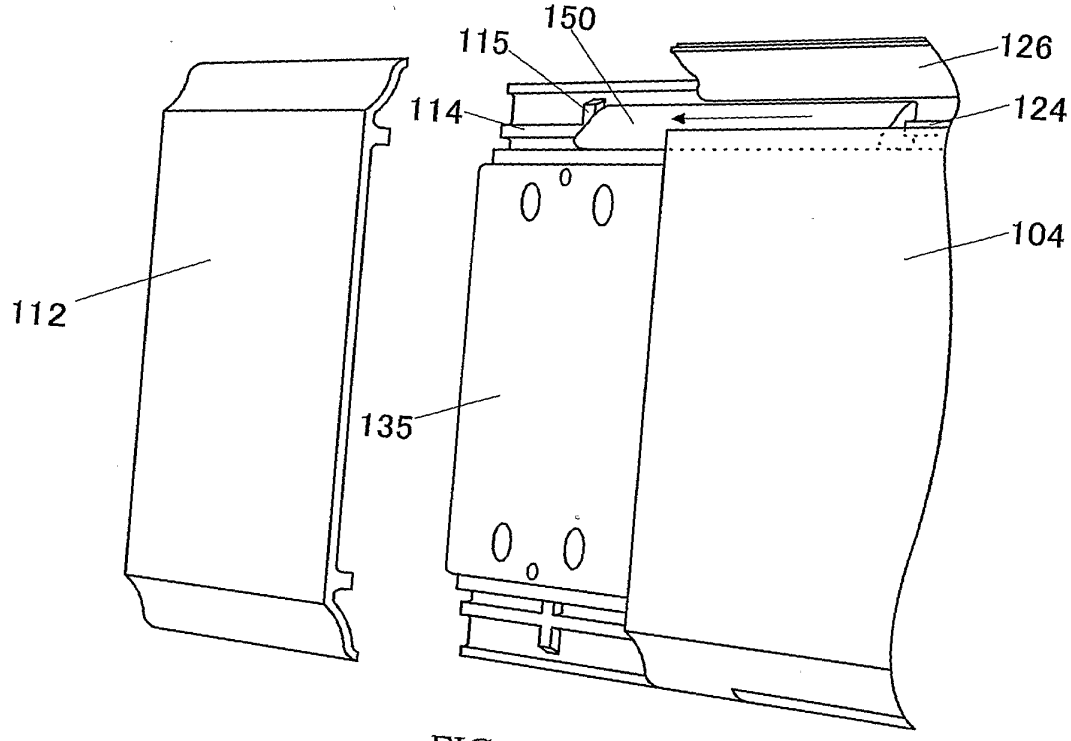


FIG. 33

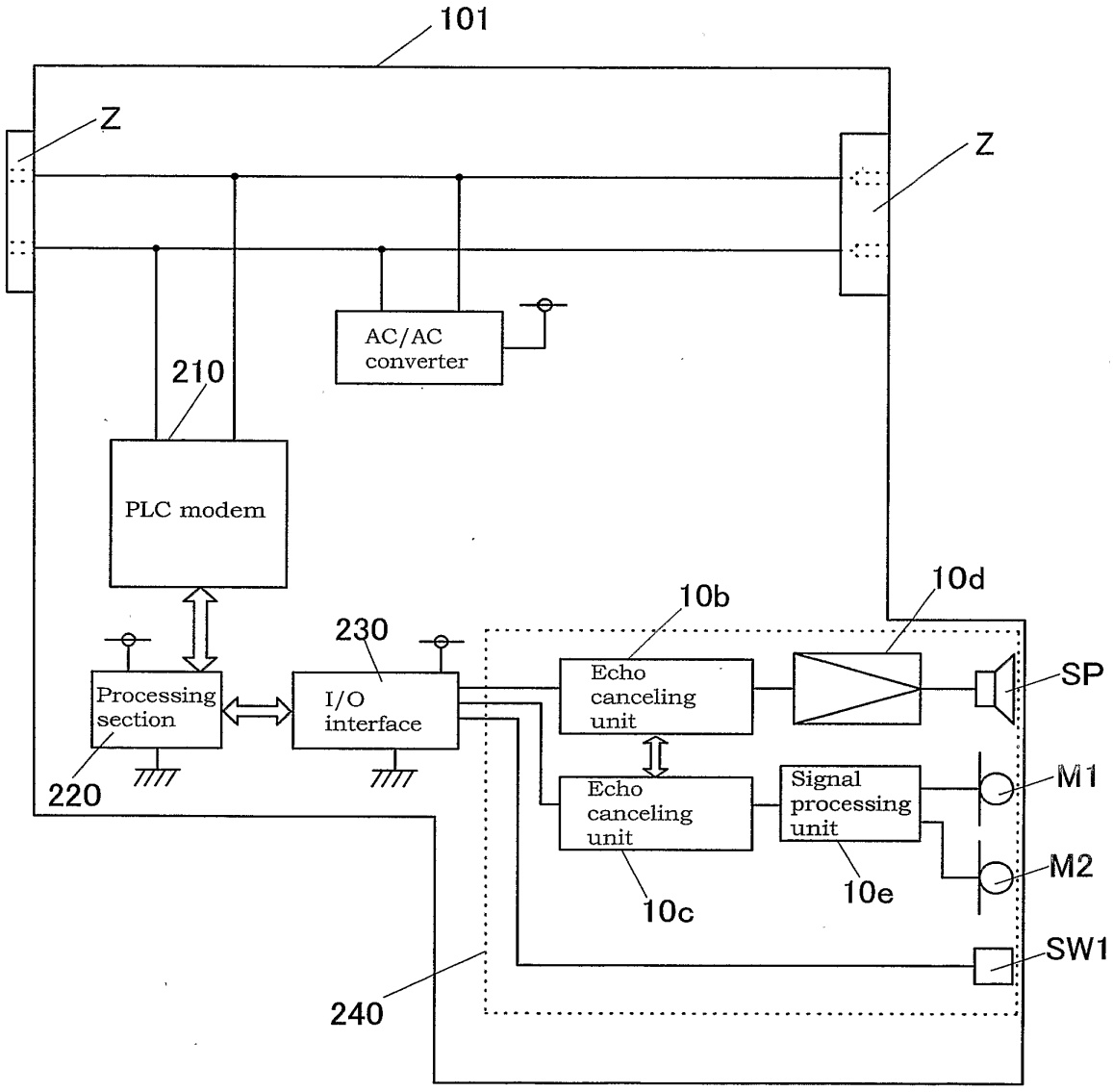

101
Z
Z
AC/AC converter
210
PLC modem
230
10b
10d
Processing section
I/O interface
Echo canceling unit
SP
220
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1
240

FIG. 34

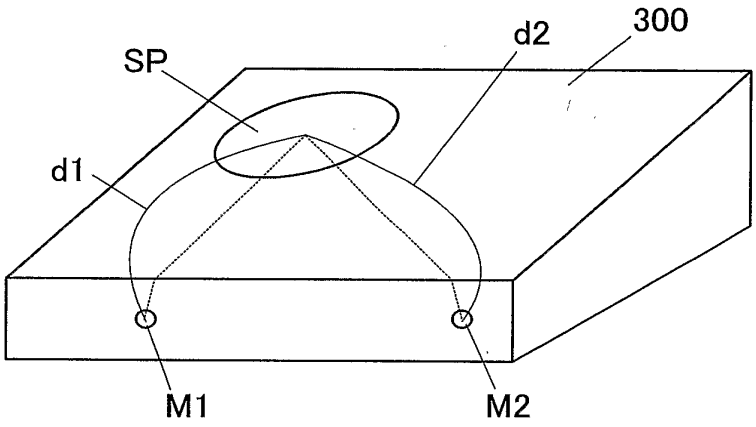


FIG. 35A

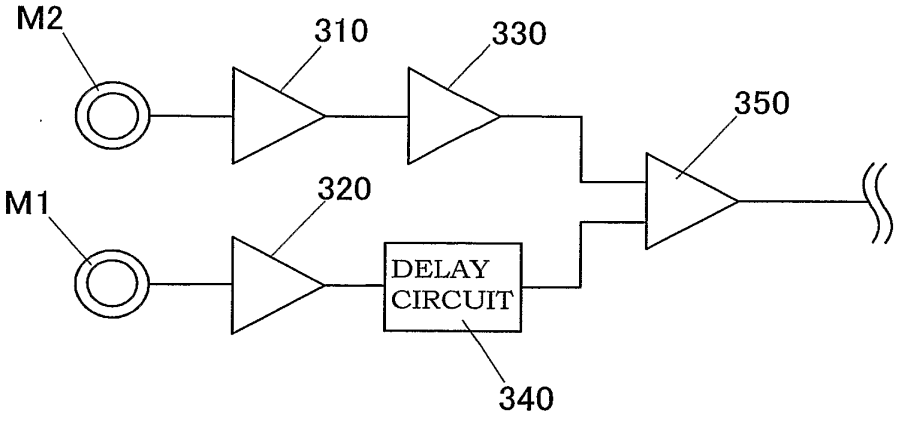


FIG. 35B

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP2007/069131

A. CLASSIFICATION OF SUBJECT MATTER

Int.Cl. H04R3/02(2006.01)i, H04M1/62(2006.01)i, H04R1/02(2006.01)i, H04R1/28(2006.01)i

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

Int.Cl. H04R3/02, H04M1/62, H04R1/02, H04R1/28

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Published examined utility model applications of Japan 1922-1996
Published unexamined utility model applications of Japan 1971-2007
Registered utility model specifications of Japan 1996-2007
Published registered utility model applications of Japan 1994-2007

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	JP 63-135100 A (BIBA KABUSHIKI KAISHA) 1988.06.07, Full Text; Fig.1 (No Family)	1-19
A	JP 2003-143677 A (MITSUBISHI ELECTRIC CORPORATION) 2003.05.16, Full Text; Fig.2 (No Family)	1-19
A	JP 01-253352 A (MATSUSHITA ELECTRIC WORKS,LTD.) 1989.10.09, Full Text; Fig.2 (No Family)	1-19

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☐ See patent family annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
25.12.2007	08.01.2008

Name and mailing address of the ISA/JP	Authorized officer
Japan Patent Office 3-4-3, Kasumigaseki, Chiyoda-ku, Tokyo 100-8915, Japan	Hiroshi Ono 5Z 3654 Telephone No. +81-3-3581-1101 Ext. 3541

Form PCT/ISA/210 (second sheet) (April 2007)

Electronic Patent Application Fee Transmittal

Application Number:	17736180
Filing Date:	04-May-2022
Title of Invention:	PERSONALIZED SOUND MANAGEMENT AND METHOD
First Named Inventor/Applicant Name:	Steven Goldstein
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-17203731

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
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($)
Miscellaneous:				
SUBMISSION- INFORMATION DISCLOSURE STMT	2806	1	130	130
		Total in USD ($)		**130**

Electronic Acknowledgement Receipt	
EFS ID:	45872575
Application Number:	17736180
International Application Number:	
Confirmation Number:	5894
Title of Invention:	PERSONALIZED SOUND MANAGEMENT AND METHOD
First Named Inventor/Applicant Name:	Steven Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-17203731
Receipt Date:	03-JUN-2022
Filing Date:	04-MAY-2022
Time Stamp:	15:15:11
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$130
RAM confirmation Number	E202263F16480868
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Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARWARE-IDS.pdf	1036831 3a48917160a6fe962b9436b5f9e61c4044fbcb85	no	11
Warnings:					
Information:					
3	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS.pdf	1037670 2bd66e75b8178bf8efc51e8a5e336dbea3d14773	no	12
Warnings:					
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4	Foreign Reference	EP1385324A1.pdf	291998 fd44f66dc1543bbf582c8d89c4cae777488fccff	no	15
Warnings:					
Information:					
5	Foreign Reference	EP1401240A1.pdf	269098 415e812ff51bde7a5bbcfea0ddc74575071e1f22	no	14
Warnings:					
Information:					
6	Foreign Reference	EP1519625B1.pdf	1060783 52f1291b06313730594dd459696b6ec0158d4382	no	27
Warnings:					
Information:					

7	Foreign Reference	WO2004114722A1.pdf	725963 bf5581f79d06ef7598680ed66e6f82a713bc0100	no	19
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Information:					
8	Foreign Reference	WO2006037156A1.pdf	1531590 348f0530551347acbea6bc41f27261c87be88ec1	no	37
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9	Foreign Reference	WO2007092660A1.pdf	159414 f1e779a0a3e51b76d6a7b22a5bb118212168a792	no	6
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Information:					
10	Foreign Reference	WO2008050583A1.pdf	2971807 88026f6ed1c5e16337df4097ac505bcea35770b2	no	67
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Information:					
11	Fee Worksheet (SB06)	fee-info.pdf	37615 dd3ebce969d6d17a3ab5fd4bd37ad319d7d406f6	no	2
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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	17/736,180	CONFIRMATION NO. 5894
APPLICATION OF	:	STEVEN GOLDSTEIN	
DATE FILED	:	4 MAY 2022	
EXAMINER	:	UNKNOWN	
TC/A.U.	:	2654	
DOCKET	:	CONT-17203731	
TITLE	:	PERSONALIZED SOUND MANAGEMENT AND METHOD	

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: June 3, 2022

Sir:

The attached extensive IDSs contain references filed in recent IPRs for similar inventive scope and similar references filed in invalidity contentions filed in Court for similar inventive scope. In the abundance of due diligence, Applicant has compiled the extensive list for the Examiner's consideration and apologizes ahead of time if any of the references are copies of previously considered related art.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,
TECHIYA LLC

Date: June 3, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
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(Not for submission under 37 CFR 1.99)

Application Number	17736180
Filing Date	2022-05-04
First Named Inventor	Steven Goldstein
Art Unit	2654
Examiner Name	
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	
	17	8140325		2012-03-20	Kanevsky	
	18	8983081		2015-03-17	Bayley	
	19	5327506		1994-07-05	Stites	

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	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	20	5933510		1999-08-03	Bryant	
	21	8086093		2011-12-27	Stuckman	
	22	7039585		2006-05-02	Wilmot	
	23	6606598		2003-08-12	Holthouse	
	24	6005525		1999-12-21	Kivela	
	25	RE38351		2003-12-16	Iseberg	
	26	7903825		2011-03-08	Melanson	
	27	8014553		2011-09-06	Radivojevic	
	28	8160261		2012-04-17	Schulein	
	29	6639987		2003-10-28	McIntosh	
	30	5923624		1999-07-13	Groeger	

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	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	31	8160273		2012-04-17	Visser	
	32	7464029		2008-12-09	Visser	
	33	7983907		2011-07-19	Visser	
	34	8254591		2012-08-28	Goldstein	
	35	8718305		2014-05-06	Usher	
	36	8798278		2014-08-05	Isabelle	
	37	8917894		2014-12-23	Goldstein	
	38	8625818		2014-01-07	Goldstein	
	39	9491542		2016-11-08	Usher	
	40	8218784		2012-07-10	Schulein	
	41	6118877		2000-09-12	Lindemann	

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	42	7050592		2006-05-23	Iseberg	
	43	8401200		2013-03-19	Tiscareno	
	44	9196247		2015-11-24	Harada	
	45	7574917		2009-08-18	Von Dach	

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	1	20030165246		2003-09-04	Kvaloy et al.	
	2	20080037801		2008-02-14	Alves et al.	
	3	20060204014		2006-09-14	Isenberg et al.	
	4	20050078838		2005-04-14	Simon	
	5	20070036377		2007-02-15	Stirnemann	

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	6	20020165719		2002-11-07	Wang	
	7	20060140425		2006-06-29	Berg	
	8	20030152359		2003-08-14	Kim	
	9	20070003090		2007-01-04	Anderson	
	10	20030165319		2003-09-04	Barber	
	11	20070255435		2007-11-01	Cohen	
	12	20040264938		2004-12-30	Felder	
	13	20060264176		2006-11-23	Hong	
	14	20040109579		2004-07-10	Izuchi	
	15	20060167687		2006-07-27	Kates	
	16	20060083390		2006-04-20	Kaderavek	

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	17	20050068171		2005-03-31	Kelliher	
	18	20050028212		2005-02-03	Laronne	
	19	20080145032		2008-06-19	Lindroos	
	20	20020098878		2002-07-25	Mooney	
	21	20060188075		2006-08-24	Peterson	
	22	20140023203		2014-01-23	Rotschild	
	23	20060188105		2006-08-24	Baskerville	
	24	20020076057		2002-06-20	Voix	
	25	20110116643		2011-05-19	Tiscareno	
	26	20150215701		2015-07-30	Usher	
	27	20140163976		2014-06-12	Park	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	28	20080137873		2008-06-12	Goldstein		
	29	20090122996		2009-05-14	Klein		
	30	20040086138		2004-05-06	Kuth		

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	1	2006054698	WO			Murata		
	2	EP1401240	EP			Jensen		
	3	2008050583	WO			Kitada		
	4	2007092660	WO			Moberly		
	5	2004114722	WO			Pedersen		
	6	EP1385324	EP			Pedersen		

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	7	2009023784	WO			Usher		
	8	2012097150	WO			Usher		

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	Application Number	17736180
	Filing Date	2022-05-04
	First Named Inventor	Steven Goldstein
	Art Unit	2654
	Examiner Name	
	Attorney Docket Number	CONT-17203731

CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-06-03
Name/Print	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

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5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

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9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
17/736,180	05/04/2022	3654	830	CONT-17203731	17	1

CONFIRMATION NO. 5894

FILING RECEIPT

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

OC000000133722089

Date Mailed: 05/10/2022

Receipt is acknowledged of this non-provisional utility patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF FIRST INVENTOR, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection.

Please verify the accuracy of the data presented on this receipt. If an error is noted on this Filing Receipt, please submit a written request for a corrected Filing Receipt, including a properly marked-up ADS showing the changes with strike-through for deletions and underlining for additions. If you received a "Notice to File Missing Parts" or other Notice requiring a response for this application, please submit any request for correction to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections provided that the request is grantable.

Inventor(s)

Steven Goldstein, Delray Beach, FL;
John P. Keady, Fairfax Station, VA;

Applicant(s)

Staton Techiya LLC, Delray Beach, FL

Assignment For Published Patent Application

Staton Techiya LLC, Delray Beach, FL

Power of Attorney: The patent practitioners associated with Customer Number 150139

Domestic Priority data as claimed by applicant

This application is a CON of 17/203,731 03/16/2021
which is a CON of 16/671,689 11/01/2019 PAT 10997978
which is a CON of 14/846,994 09/07/2015 PAT 10529325
which is a CON of 12/560,097 09/15/2009 PAT 9129291
which claims benefit of 61/098,914 09/22/2008

Foreign Applications for which priority is claimed (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.) - None.
Foreign application information must be provided in an Application Data Sheet in order to constitute a claim to foreign priority. See 37 CFR 1.55 and 1.76.

Permission to Access Application via Priority Document Exchange: Yes

Permission to Access Search Results: Yes

Applicant may provide or rescind an authorization for access using Form PTO/SB/39 or Form PTO/SB/69 as appropriate.

If Required, Foreign Filing License Granted: 05/09/2022
The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 17/736,180**
Projected Publication Date: 08/18/2022
Non-Publication Request: No
Early Publication Request: No
**** SMALL ENTITY ****
Title

PERSONALIZED SOUND MANAGEMENT AND METHOD

Preliminary Class

474

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications: No

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Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

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Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative,

this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4258).

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Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

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The grant of a license does not in any way lessen the responsibility of a licensee for the security of the subject matter as imposed by any Government contract or the provisions of existing laws relating to espionage and the national security or the export of technical data. Licensees should apprise themselves of current regulations especially with respect to certain countries, of other agencies, particularly the Office of Defense Trade Controls, Department of State (with respect to Arms, Munitions and Implements of War (22 CFR 121-128)); the Bureau of Industry and Security, Department of Commerce (15 CFR parts 730-774); the Office of Foreign AssetsControl, Department of Treasury (31 CFR Parts 500+) and the Department of Energy.

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PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 17/736,180

APPLICATION AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	80		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	350		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	400		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	17 minus 20 =	*	x 50 =	0.00	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	1 minus 3 =	*	x 240 =	0.00			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	830		TOTAL	

APPLICATION AS AMENDED - PART II

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT A	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	RATE($)	ADDITIONAL FEE($)	OR	RATE($)	ADDITIONAL FEE($)
AMENDMENT B	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest found in the appropriate box in column 1.

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	
Filing Date	
First Named Inventor	Goldstein
Art Unit	
Examiner Name	
Attorney Docket Number	CONT-17203731

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	9129291		2015-09-08	Goldstein; Steven	
	2	10529325		2020-01-07	Goldstein; Steven	
	3	6661901		2003-12-09	Svean; Jarle	
	4	6754359		2004-06-22	Svean; Jarle	
	5	8208644		2012-06-26	Goldstein; Steven W.	
	6	8315400		2012-11-20	Goldstein; Steven Wayne	
	7	9781530		2017-10-03	Usher; John	
	8	10097149		2018-10-09	Voix; Jeremie	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	
	Examiner Name	
	Attorney Docket Number	CONT-17203731

	9	9191740		2015-11-17	McIntosh et al.	
	10	3902155		1975-08-26	Young	
	11	4539440		1985-09-03	Sciarra	
	12	4731850		1988-03-15	Levitt et al.	
	13	5033090		1991-07-16	Weinrich	
	14	5734961		1998-03-31	Castille	
	15	5894113		1999-04-13	Wingate	
	16	6469615		2002-10-22	Kady et al.	
	17	6606596		2003-08-12	Zimgibl et al.	
	18	8126159		2012-02-28	Goose et al.	

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	Filing Date	
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	Attorney Docket Number	CONT-17203731

Examiner Initial*	Cite No	Publication Number	Kind Code[1]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	20050254667		2005-11-17	Michael, Kevin	
	2	20080107286		2008-05-08	Nakajima; Kaori	
	3	20090220096		2009-09-03	Usher; John	
	4	20130218560		2013-08-22	Hsiao; Jen-Po	
	5	20180176696		2020-02-25	Voigt Pedersen; Soren Christian	
	6	20040179694		2004-09-16	Kenneth Alley	
	7	20100033313		2010-02-11	Keady et al.	
	8	20100232634		2010-09-16	Boldt	
	9	20120021732		2012-01-26	Lam	
	10	20020129056		2002-09-12	Conant et al.	

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	Examiner Name	
	Attorney Docket Number	CONT-17203731

	11	20020183648		2002-12-05	Hou	
	12	20030185403		2003-10-02	Sibbald	
	13	20050015606		2005-01-20	Blamires et al.	
	14	20050049492		2005-03-03	Sweeney et al.	
	15	20050098681		2005-05-12	Berson et al.	
	16	20050105750		2005-05-19	Frohlich	
	17	20050251224		2005-11-10	Berg	
	18	20050144132		2005-06-30	Lin	
	19	20060158956		2006-07-20	Laugharn et al.	
	20	20060233395		2006-10-19	Orten	
	21	20060262935		2006-11-23	Goose et al.	

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	Examiner Name	
	Attorney Docket Number	CONT-17203731

	22	20070098195		2007-05-03	Holmes	
	23	20080091605		2008-04-17	Hughes et al.	
	24	20080120115		2008-05-22	Mao	
	25	20080162133		2008-07-03	Couper et al.	
	26	20080165980		2008-07-10	Pavlovic	
	27	20080177998		2008-07-24	Apsangi et al.	
	28	20080178088		2008-07-24	Goldstein et al.	
	29	20080267422		2008-10-30	Cox	
	30	20090110223		2009-04-30	Semcken	
	31	20090183560		2009-07-23	Henne et al.	
	32	20090190767		2009-07-30	Aaron et al.	

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	Examiner Name	
	Attorney Docket Number	CONT-17203731

	33	20090204243		2009-08-13	Marwaha et al.	
	34	20100002897		2010-01-07	Keady	

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Examiner Initial*	Cite No	Foreign Document Number[3]	Country Code[2]	Kind Code[4]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T[5]
	1	CN101227759	CN		2008-07-23			
	2	1104155	EP		2011-05-30	Phone Com Inc.		
	3	WO0028721	WO		2000-05-18	Ericsson Inc.		
	4	WO08014482	WO		2008-01-31	Personics Holdings		
	5	WO08061260	WO		2008-05-22	Personics Holdings		
	6	WO08153589	WO		2008-12-18	Personics Holdings		

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	Attorney Docket Number	CONT-17203731

Examiner Initials*	Cite No	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc), date, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]
	1	Anonymous, Choose your flavor: Sound ID launches Two new Bluetooth Headsets with Personalized Sound, Business Wire, New York: Oct. 16, 2006. pp. 1-4. Doc ID 1146 19641	
	2	PCT Internationa1 Search Report for PCT Application No. PCT/US07/74641 dated Jul. 17 2008	
	3	PCT International Search Report for PCT Application No. PCT!US07/85134 dated Jun. 12 2008	
	4	PCT International Search Report for PCT Application No. PCT!US07/85134 dated Jun. 12 2008	
	5	PCT Written Opinion for PCT Applicaiton No. PCTIUS07/85134 dated Jun. 12 2008	
	6	PCT Written Opinion for PCT Application No. PCT!US08/88417 dated A . 9 2009	
	7	Patent Cooperation Treaty, "International Search Report," issued in International Patent Application No. PCT!US09/57751, by U.S. Search Authority dated Nov. 6 2009 document of 2 pages	

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EXAMINER SIGNATURE

Examiner Signature		Date Considered	

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[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-05-02
Name/Print	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

CLAIMS

We Claim:

1. An Always On Recording system comprising:

a user interface; and

an audio device comprising:

a microphone configured to measure ambient sound and generate a microphone signal;

a speaker configured to emit an audio signal;

a memory configured to store instructions;

an audio buffer configured to store at least a portion of the microphone signal;

a data buffer;

a processor operatively coupled to the memory, the processor operatively coupled to the audio buffer, the processor operatively coupled to the data buffer, wherein the processor is configured to execute the instructions to perform operations comprising:

receiving the microphone signal;

sending a portion of the microphone signal to the audio buffer;

analyzing the audio buffer for detecting a sonic signature;

notifying the user when the sonic signature is detected;

sending a modified microphone signal to the data buffer replacing or adding to previously data stored in the data buffer; and

sending a portion of the data buffer, spanning a time period, to the speaker in response to a user request, wherein the user request is one of the sonic signature that is a voice command or a manual input from the user interface.

2. The system according to claim 1, wherein the sonic signature is a voice command, and where the voice command is to replay audio.

3. The system according to claim 1, wherein the time period is less than 30 minutes.

4. The system according to claim 1, wherein the time period is less than 15 minutes.

5. The system according to claim 1, wherein the audio device is wirelessly connected to an earpiece.

6. The system according to claim 5, wherein the earpiece includes:

an ambient sound microphone; and

an ear canal receiver.

7. The system according to claim 6, wherein the earpiece further includes:

an ear canal microphone.

8. The system according to claim 7, wherein the operation of notifying the user sends a notification signal to the ear canal receiver.

9. The system according to claim 1, wherein the sonic signature detected is one of a siren or an alarm.

10. The system according to claim 9 wherein the earpiece further includes:

a sealing section.

11. The system according to claim 9 wherein the user interface is a button.

12. The system according to claim 1, wherein the time period is less than 1 minute.

Electronic Patent Application Fee Transmittal

Application Number:	
Filing Date:	
Title of Invention:	PERSONALIZED SOUND MANAGEMENT AND METHOD
First Named Inventor/Applicant Name:	Steven Goldstein
Filer:	John Patrick Keady
Attorney Docket Number:	CONT-17203731

Filed as Small Entity

Filing Fees for Utility under 35 USC 111(a)

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
UTILITY FILING FEE (ELECTRONIC FILING)	4011	1	80	80
UTILITY SEARCH FEE	2111	1	350	350
UTILITY EXAMINATION FEE	2311	1	400	400
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**830**

Electronic Acknowledgement Receipt	
EFS ID:	45629868
Application Number:	17736180
International Application Number:	
Confirmation Number:	5894
Title of Invention:	PERSONALIZED SOUND MANAGEMENT AND METHOD
First Named Inventor/Applicant Name:	Steven Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT-17203731
Receipt Date:	04-MAY-2022
Filing Date:	
Time Stamp:	10:56:35
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	CARD
Payment was successfully received in RAM	$830
RAM confirmation Number	E202254A58371360
Deposit Account	
Authorized User	
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Application Data Sheet	WebADS.pdf	206636 24d85fb534051d8a01b3d3cf5f0a15a1b027b2fb	no	8
Warnings:					
Information:					
2	Specification	CONT-17203731-spec.pdf	218474 6538acc3365e798bd5d872a39b568f6ceb02d0e8	no	25
Warnings:					
Information:					
3	Abstract	CONT-17203731-abstract.pdf	76446 17c3983232014780ee1ca6db142f143f82c57e66	no	1
Warnings:					
Information:					
4	Drawings-only black and white line drawings	CONT-17203731_figures.pdf	809208 98e8ce735eca0cc36e3bc7393599c3d0c60aa565	no	15
Warnings:					
Information:					
5	Transmittal Letter	CONT-17203731_TRANSMTTL-LTR.pdf	135309 64a6ade9ecda3caaac82deb1b90bd8eba5842fba	no	5
Warnings:					
Information:					
6	Transmittal Letter	CONT-17203731-IDS-transmittal-ltr.pdf	120457 d353190a039ee83cb355db0cfcb6201891f29061	no	1
Warnings:					
Information:					

7	Power of Attorney	11472-15-POA-Executed.pdf	70455 467bd95e15a395c9129e6d558ac065f73c2692a2	no	1
Warnings:					
Information:					
8	Oath or Declaration filed	CONT-17203731_oath_PARENT.pdf	476122 6a8498a2dd149dfa136d12b590b7844dafeebbfc	no	3
Warnings:					
Information:					
9	Information Disclosure Statement (IDS) Form (SB08)	IPR-IDS.pdf	1034947 6ca73d5a94d91906217 8f8511924 7499d152bee6	no	5
Warnings:					
Information:					
A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
10	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARWARE-IDS.pdf	1035272 4bbb3f552c3760bcc54150bf9667f6b8e00b5630	no	6
Warnings:					
Information:					
11	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS.pdf	1036959 ca86c5f1899464658de09d310cb5269bec9b4b73	no	10
Warnings:					
Information:					
12	Information Disclosure Statement (IDS) Form (SB08)	CONT-17203731-IDS_B.pdf	1036344 bed5bc8dc2c6b108fea178c736ef8500a340928b	no	9
Warnings:					
Information:					
13	Information Disclosure Statement (IDS) Form (SB08)	CONT-17203731-IDS_A.pdf	1036627 8d908b79c2f594b13b2581b47e3a1c8d80708d35	no	9

Warnings:					
Information:					
14	Non Patent Literature	IPR2A-reduced.pdf	1663015 b990440678e64459d2a9aa20b079329da62175c0	no	79
Warnings:					
Information:					
15	Non Patent Literature	IPR3A-reduced.pdf	1058068 0464efd1784c4771159f66a1f6e900e0133b06b0	no	80
Warnings:					
Information:					
16	Non Patent Literature	IPR5A-reduced.pdf	7320303 b752b0496de0a73a97a8b5b45885ad760da92753	no	93
Warnings:					
Information:					
17	Non Patent Literature	IPR4A-reduced.pdf	3239458 cfd74847961e1baef24212c8633212bcbd396ca5	no	88
Warnings:					
Information:					
18	Non Patent Literature	IPR6A-reduced.pdf	2876288 d7dc265d9c94073c7d3c98486ed0080222ae1f1e	no	96
Warnings:					
Information:					
19	Non Patent Literature	IPR7A-reduced.pdf	6361523 56b1b0c2ddf9bafa5116c746b4039ab606aee614	no	91
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Information:					
20	Non Patent Literature	IPR8A-reduced.pdf	907038 547aba21f3066a7033bb0b4648b05cbec7499fe7	no	79
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Information:					

21	Non Patent Literature	IPR9A-reduced.pdf	2564425 4b9d6530768eefcdcd796e31b768bef59ef13a56	no	84
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22	Non Patent Literature	IPR10A-reduced.pdf	1661463 7fab859aa25cd2aa0d190e34234b5b8be890c0da	no	58
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23	Non Patent Literature	IPR11A-reduced.pdf	1790447 3d29f05b0d3d302b325d50c3f55626b746f24b18	no	84
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Information:					
24	Non Patent Literature	IPR1A-reduced.pdf	900793 c46884c96d041e70fa09728978b8e242a34fd2ac	no	80
Warnings:					
Information:					
25	Non Patent Literature	ref2-pdf2.pdf	3762561 39646ce913fa97e4a1efb547fcce8a48fc8e6d08	no	10
Warnings:					
Information:					
26	Non Patent Literature	ref1.pdf	1115022 01c5b583e9d89f0e10dda2f41631f1a7c8c19dda	no	2
Warnings:					
Information:					
27	Non Patent Literature	ref3.pdf	2858776 2418fbe7a8c340eaa35ef3922c589767859789c3	no	3
Warnings:					
Information:					

28	Claims	CONT-17203731-claims.pdf	108392 19c4c2ab81449aab5feb3d7d1b99111cb920a727	no	3
Warnings:					
Information:					
29	Fee Worksheet (SB06)	fee-info.pdf	43884 f43a0e18f24e2ee9dc724d9c92f10966ddafd8fb	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	45524712		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

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.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

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.

Secrecy Order 37 CFR 5.2:

☐ 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.)

Inventor Information:

Inventor 1

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Steven		Goldstein	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	Delray Beach	State/Province	FL	Country of Residence i	US

Mailing Address of Inventor:

Address 1	16390 Maddalena Place		
Address 2			
City	Delray Beach	State/Province	FL
Postal Code	33446	Country i	US

Inventor 2

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	John	P.	Keady	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	Fairfax Station	State/Province	VA	Country of Residence i	US

Mailing Address of Inventor:

Address 1	8138 Crestridge Rd.		
Address 2			
City	Fairfax Station	State/Province	VA
Postal Code	22039	Country i	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

PTO/AIA/14 (08-15)
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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Enter either Customer Number or complete the Correspondence Information section below. For further information see 37 CFR 1.33(a).			
☐ **An Address is being provided for the correspondence Information of this application.**			
Customer Number	150139		
Email Address	ip@akerman.com	Add Email	Remove Email
Email Address	personics_docketing@cardinal-ip.com	Add Email	Remove Email

Application Information:

Title of the Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD		
Attorney Docket Number	CONT-17203731	**Small Entity Status Claimed**	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)	15	**Suggested Figure for Publication (if any)**	14

Filing By Reference:

Only complete this section when filing an application by reference under 35 U.S.C. 111(c) and 37 CFR 1.57(a). Do not complete this section if application papers including a specification and any drawings are being filed. Any domestic benefit or foreign priority information must be provided in the appropriate section(s) below (i.e., "Domestic Benefit/National Stage Information" and "Foreign Priority Information").

For the purposes of a filing date under 37 CFR 1.53(b), the description and any drawings of the present application are replaced by this reference to the previously filed application, subject to conditions and requirements of 37 CFR 1.57(a).

Application number of the previously filed application	Filing date (YYYY-MM-DD)	Intellectual Property Authority or Country

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 and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Either enter Customer Number or complete the Representative Name section below. If both sections are completed the customer Number will be used for the Representative Information during processing.

Please Select One:	◉ Customer Number	○ US Patent Practitioner	○ Limited Recognition (37 CFR 11.9)

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Customer Number	150139				
Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
Registration Number					
Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
Registration Number					
Additional Representative Information blocks may be generated within this form by selecting the **Add** button.					

Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, 365(c), or 386(c) or indicate National Stage entry from a PCT application. Providing benefit claim information in the Application Data Sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78.
When referring to the current application, please leave the "Application Number" field blank.

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
	Continuation of	17203731	2021-03-16

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
17203731	Continuation of	16671689	2019-11-01	10997978	2021-05-04

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
16671689	Continuation of	14846994	2015-09-07	10529325	2020-01-07

Prior Application Status	Patented				Remove
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
14846994	Continuation of	12560097	2009-09-15	9129291	2015-09-08

Prior Application Status	Pending		Remove
Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
12560097	Claims benefit of provisional	61098914	2008-09-22

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button.

Foreign Priority Information:

This section allows for the applicant to claim priority to a foreign application. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55. When priority is claimed to a foreign application that is eligible for retrieval under the priority document exchange program (PDX) the information will be used by the Office to automatically attempt retrieval pursuant to 37 CFR 1.55(i)(1) and (2). Under the PDX program, applicant bears the ultimate responsibility for ensuring that a copy of the foreign application is received by the Office from the participating foreign intellectual property office, or a certified copy of the foreign priority application is filed, within the time period specified in 37 CFR 1.55(g)(1).

Application Number	Country	Filing Date (YYYY-MM-DD)	Access Code (if applicable)
	US		

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button.

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications

☐ This application (1) claims priority to or the benefit of an application filed before March 16, 2013 and (2) also contains, or contained at any time, a claim to a claimed invention that has an effective filing date on or after March 16, 2013.
NOTE: By providing this statement under 37 CFR 1.55 or 1.78, this application, with a filing date on or after March 16, 2013, will be examined under the first inventor to file provisions of the AIA.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Authorization or Opt-Out of Authorization to Permit Access:

When this Application Data Sheet is properly signed and filed with the application, applicant has provided written authority to permit a participating foreign intellectual property (IP) office access to the instant application-as-filed (see paragraph A in subsection 1 below) and the European Patent Office (EPO) access to any search results from the instant application (see paragraph B in subsection 1 below).

Should applicant choose not to provide an authorization identified in subsection 1 below, applicant **must opt-out** of the authorization by checking the corresponding box A or B or both in subsection 2 below.

NOTE: This section of the Application Data Sheet is **ONLY** reviewed and processed with the **INITIAL** filing of an application. After the initial filing of an application, an Application Data Sheet cannot be used to provide or rescind authorization for access by a foreign IP office(s). Instead, Form PTO/SB/39 or PTO/SB/69 must be used as appropriate.

1. Authorization to Permit Access by a Foreign Intellectual Property Office(s)

A. Priority Document Exchange (PDX) - Unless box A in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the European Patent Office (EPO), the Japan Patent Office (JPO), the Korean Intellectual Property Office (KIPO), the State Intellectual Property Office of the People's Republic of China (SIPO), the World Intellectual Property Organization (WIPO), and any other foreign intellectual property office participating with the USPTO in a bilateral or multilateral priority document exchange agreement in which a foreign application claiming priority to the instant patent application is filed, access to: (1) the instant patent application-as-filed and its related bibliographic data, (2) any foreign or domestic application to which priority or benefit is claimed by the instant application and its related bibliographic data, and (3) the date of filing of this Authorization. See 37 CFR 1.14(h)(1).

B. Search Results from U.S. Application to EPO - Unless box B in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the EPO access to the bibliographic data and search results from the instant patent application when a European patent application claiming priority to the instant patent application is filed. See 37 CFR 1.14(h)(2).

The applicant is reminded that the EPO's Rule 141(1) EPC (European Patent Convention) requires applicants to submit a copy of search results from the instant application without delay in a European patent application that claims priority to the instant application.

2. Opt-Out of Authorizations to Permit Access by a Foreign Intellectual Property Office(s)

☐ A. Applicant **DOES NOT** authorize the USPTO to permit a participating foreign IP office access to the instant application-as-filed. If this box is checked, the USPTO will not be providing a participating foreign IP office with any documents and information identified in subsection 1A above.

☐ B. Applicant **DOES NOT** authorize the USPTO to transmit to the EPO any search results from the instant patent application. If this box is checked, the USPTO will not be providing the EPO with search results from the instant application.

NOTE: Once the application has published or is otherwise publicly available, the USPTO may provide access to the application in accordance with 37 CFR 1.14.

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Approved for use through 04/30/2017. OMB 0651-0032
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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Applicant Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Applicant 1

If the applicant is the inventor (or the remaining joint inventor or inventors under 37 CFR 1.45), this section should not be completed. The information to be provided in this section is the name and address of the legal representative who is the applicant under 37 CFR 1.43; or the name and address of the assignee, person to whom the inventor is under an obligation to assign the invention, or person who otherwise shows sufficient proprietary interest in the matter who is the applicant under 37 CFR 1.46. If the applicant is an applicant under 37 CFR 1.46 (assignee, person to whom the inventor is obligated to assign, or person who otherwise shows sufficient proprietary interest) together with one or more joint inventors, then the joint inventor or inventors who are also the applicant should be identified in this section.

Clear

○ Assignee	○ Legal Representative under 35 U.S.C. 117	○ Joint Inventor
◉ Person to whom the inventor is obligated to assign.	○ Person who shows sufficient proprietary interest	

If applicant is the legal representative, indicate the authority to file the patent application, the inventor is:

Name of the Deceased or Legally Incapacitated Inventor:	
If the Applicant is an Organization check here.	☒
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Additional Applicant Data may be generated within this form by selecting the Add button.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Assignee Information including Non-Applicant Assignee Information:

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Assignee 1

Complete this section if assignee information, including non-applicant assignee information, is desired to be included on the patent application publication. An assignee-applicant identified in the "Applicant Information" section will appear on the patent application publication as an applicant. For an assignee-applicant, complete this section only if identification as an assignee is also desired on the patent application publication.

If the Assignee or Non-Applicant Assignee is an Organization check here. ☒

Organization Name	Staton Techiya LLC

Mailing Address Information For Assignee including Non-Applicant Assignee:

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Country i	US	Postal Code	33446
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Additional Assignee or Non-Applicant Assignee Data may be generated within this form by selecting the Add button.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT-17203731
	Application Number	
Title of Invention	PERSONALIZED SOUND MANAGEMENT AND METHOD	

Signature:

NOTE: This Application Data Sheet must be signed in accordance with 37 CFR 1.33(b). **However, if this Application Data Sheet is submitted with the INITIAL filing of the application and either box A or B is not checked in subsection 2 of the "Authorization or Opt-Out of Authorization to Permit Access" section, then this form must also be signed in accordance with 37 CFR 1.14(c).**

This Application Data Sheet **must** be signed by a patent practitioner if one or more of the applicants is a **juristic entity** (e. g., corporation or association). If the applicant is two or more joint inventors, this form must be signed by a patent practitioner, **all** joint inventors who are the applicant, or one or more joint inventor-applicants who have been given power of attorney (e.g., see USPTO Form PTO/AIA/81) on behalf of **all** joint inventor-applicants.

See 37 CFR 1.4(d) for the manner of making signatures and certifications.

Signature	/John P. Keady/			Date (YYYY-MM-DD)	
First Name	John	Last Name	Keady	Registration Number	56389

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PERSONALIZED SOUND MANAGEMENT AND METHOD

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of and claims priority benefit to U.S. Application Ser. No. 17/203,731, filed 16 Mar. 2021, which is a continuation of U.S. Application Ser. No. 16/671,689, filed 1 Nov. 2019, which is a continuation of U.S. application Ser. No. 14/846,994, filed 7 Sep. 2015. U.S. application Ser. No. 14/846,994 is a continuation of U.S. application Ser. No. 12/560,097 filed on 15 Sep. 2009 and claims the benefit of U.S. provisional patent application No. 61/098,914 filed 22 Sep. 2008. The disclosure of the aforementioned applications are all incorporated herein by reference in their entireties.

FIELD OF THE INVENTION

[0002] The invention relates in general to methods of managing sound, and particularly though not exclusively, is related to personalized sound management.

BACKGROUND OF THE INVENTION

[0003] The world of two hundred years ago is substantially different than the present day earth. Similarly, the acoustic environment that surrounds us is also changing. For example, the sounds of a large city have changed as the mode of transportation transitioned from walking and horse and buggy, to cars, subways, and airplanes.

[0004] In general, humans are continuously inundated by a diversity of sounds. Many of the sounds are not critical to our lives but our brain processes these sounds and tries to distinguish between them. Background sound levels can also make it difficult to hear sounds that are important. Too much acoustic information can cause an auditory overload that can impact both the health and safety of an individual.

SUMMARY

[0005] The invention relates in general to methods and system for implementing a suite of personalized sound applications for modifying a user's acoustic environment and more particularly, though not exclusively, to facilitating the adoption of the technology, ensuring the technology functions properly, protecting both the manufacturers and consumer, and providing user selection and control over the management of sound.

[0006] At least one exemplary embodiment is directed to a method of personalized sound management comprising the steps of: selecting at least one of a plurality of personalized sound management applications through a client system where the user selects the at least one of the plurality of personalized sound management applications from a website; accepting a subscription contract for using the at least one of the personalized sound management applications; and loading the selected at least one of the plurality of applications from a server system to a device where the device has at least one microphone, at least one speaker, and a processor configured to identify sonic signatures where each sonic signature is identified using a Gaussian mixture model.

[0007] At least one exemplary embodiment is directed to a method of implementing personalized sound management comprising the steps of: recording sound with a microphone of a communication device; analyzing the sound for acoustic information relevant for personalized sound management applications; storing a sonic signature in a memory of the communication device; calculating a sound pressure level of the sonic signature; and attaching and storing metadata related to the sonic signature and sound pressure level including a time stamp and geocode.

BRIEF DESCRIPTION OF THE DRAWINGS

[0008] Exemplary embodiments of the present invention will become more fully understood from the detailed description and the accompanying drawings, wherein:

[0009] FIG. 1 illustrates a block diagram of the interaction of personalized sound management in accordance with at least one exemplary embodiment;

[0010] FIG. 2 illustrates a block diagram of a partial list of applications for personalized sound management in accordance with at least one exemplary embodiment;

[0011] FIG. 3 illustrates a block diagram of the interaction of a personalized management enabled device with other devices in accordance with at least one exemplary embodiment;

[0012] FIG. 4 illustrates a flow chart of a process of providing personalized sound management in accordance with at least one exemplary embodiment;

[0013] FIG. 5 illustrates a flow chart of an application testing in accordance with at least one exemplary embodiment;

[0014] FIG. 6 illustrates a flow chart of the testing of a personal sound management device in accordance with at least one exemplary embodiment;

[0015] FIGS. 7a and 7b are diagrams illustrating a consumer purchase process in accordance with at least one exemplary embodiment;

[0016] FIG. 8 illustrates a flow chart of registering a new device that includes personalized sound management applications in accordance with at least one exemplary embodiment;

[0017] FIG. 9 illustrates a flow chart of enabling the new device in accordance with at least one exemplary embodiment;

[0018] FIG. 10 illustrates a flow chart of updating a unit or device in accordance with at least one exemplary embodiment;

[0019] FIG. 11 illustrates a diagram of a device for implementing personalized sound management in

accordance with at least one exemplary embodiment;

[0020] FIG. 12 illustrates a block diagram of a device for implementing personalized sound management in accordance with at least one exemplary embodiment;

[0021] FIG. 13 illustrates a diagram of a communication device or earpiece configured to provide sonic signatures to a sonic signature database in accordance with at least one exemplary embodiment; and

[0022] FIG. 14 illustrates a block diagram of a cell phone capturing a sonic signature and providing the sonic signature to a database of sounds in accordance with at least one exemplary embodiment.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS OF THE PRESENT INVENTION

[0023] The following description of exemplary embodiment(s) is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0024] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the art may not be discussed in detail but are intended to be part of the enabling description where appropriate. For example specific computer code may not be listed for achieving each of the steps discussed, however one of ordinary skill would be able, without undo experimentation, to write such code given the enabling disclosure herein. Such code is intended to fall within the scope of at least one exemplary embodiment.

[0025] Additionally, the sizes of structures used in exemplary embodiments are not limited by any discussion herein (e.g., the sizes of structures can be macro (centimeter, meter, millimeter, micro (micro meter), nanometer size and smaller).

[0026] Notice that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed or further defined in the following figures.

[0027] In all of the examples illustrated and discussed herein, any specific values, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0028] FIG. 1 is a diagram of personal sound management in accordance with at least one exemplary embodiment. In our global community 100, each person is unique. The differences between individuals can be genetic, cultural, environmental, personal, or physical to name just a few. The combination of these traits is what makes us unique. One of the main senses a human relies on is our auditory sense. Hearing impacts every aspect of our life from communication to safety. How each individual perceives and uses sound is also unique.

[0029] From an acoustic perspective, sound pressure levels have been rising steadily. The diversity of sounds is also increasing. The human brain continuously processes the acoustic information provided by

the ears. Both the sound pressure levels and the sound diversity puts increasing strain on a person to determine what they need to hear versus what they are hearing. Ultimately, this "acoustic overload" can manifest itself in physical ailments and health risks such as stress, sleeplessness, and depression.

[0030] A person 102 has an acoustic space 104 from which a majority of sounds they receive emanate. Some of the sounds are useful to person 102 and some sounds may have no use. Acoustic space 104 can be large or small. Acoustic space 104 will change with time and location. For example, acoustic space 104 can be a room, a stadium, a forest, an automobile, a plane, or the ear canal of person 102. Personalized sound management (PSM™) 106 is the ability to modify what is received from acoustic space 104 thereby tailoring or personalizing the received acoustic information to meet the needs of person 102.

[0031] Devices 108 are a source of acoustic information within acoustic space 104. In general, device(s) 108 have a speaker, a microphone, or one or more of both. In general, application hardware and software 110 is incorporated in, although can be external also, device(s) 108 to allow personalization of acoustic space 104. Person 102 selects the applications and controls device(s) 108 to modify acoustic space 104 to meet their personal needs and wants. The benefit and utility of personalizing and managing sound received by person 102 will be disclosed in more detail.

[0032] FIG. 2 is a diagram illustrating a partial list of applications for personalized sound management in accordance with at least one exemplary embodiment. A user 202 has an acoustic space 204 that includes both real sounds and potential sounds. In many cases, acoustic space 204 can be inundated with a variety of sounds not under control of user 202. This lack of control over acoustic information and sound within acoustic space 204 can make for a reduction in quality of life, loss of efficiency, or more seriously affecting user's 202 health and safety.

[0033] In at least one exemplary embodiment, a user 202 selects 201 at least one personalized sound management application from a personalized sound management applications list 206. Although only a partial list, applications such as Safe Space™ 216, EarGuard™ 218, Quiet Call™ 220, Ear Sticky™ 230, Hearable™ 222, Always On-Recording™ 224, Earbliss™ 226, Hear & Tell™ 232, and Ear Mail™ 228 will be used to illustrate how acoustic space 204 is personalized and managed for the benefit of user 202. A more detailed explanation of personalized sound management applications 206 will be described.

[0034] The selected applications of user 202 are loaded into ROM 212. An operating system 208 is configured operably with device 210 and ROM 212. Operating system 208, in conjunction with personalized sound management applications 206, provides an interface to the user for personalized sound management 214 to meet user needs to manage user acoustic space 202 and for controlling device 210.

[0035] ROM 212 can be read only memory such that the selected applications from personalized sound management list 206 cannot be tampered with or rewritten. However in at least one exemplary embodiment ROM 212 can also be memory that can be read and written to so that a user can change settings. Alternately, ROM 212 can be other types of memory such as fe-ram, phase change memory, magneto-resistive memory, hard drive, sram, dram, eprom, eeprom, and other non-volatile or read only

memories where the selected applications are secure from any type of downloading, tampering, or modification through physical or software protection. Monitoring of ROM 212 and the selected applications can also be added where notification is sent or device 210 is disabled when an improper action is detected. ROM 212 ensures that an application will operate as disclosed for user 202.

[0036] Components of device 210 can be built in a single unit or operatively coupled units. For example, multiple devices of 210 can be wired, optically connected, wirelessly connected or a combination thereof. Operating system 208 can be run remotely to operate device 210 or reside within device 210. Similarly, ROM 212 can be located in device 210 or remote to device 210. In at least one exemplary embodiment, operating system 208 resides in ROM 212. Device 210 typically has at least one microphone or at least one speaker, or both. The at least one microphone provides acoustic information for use in conjunction with personalized sound management applications. The at least one speaker provides acoustic information to user 202 and acoustic space 204. Device 210 has a microprocessor (not shown) for running applications 206. In at least one exemplary embodiment the microprocessor is dedicated to running personalized sound management applications 206.

[0037] In general, personalized sound management applications are customizable point solutions that allow user 202 to handle a variety of tasks associated with managing acoustic space 204. The selection of personalized sound management applications will depend on the person, the device, and the acoustic space being managed. Non-limiting examples of devices that can use personalized sound management applications with the appropriate hardware and operating system, are earpieces, media devices, and vehicles.

[0038] Described hereinbelow are brief overviews of the user selected personalized sound management applications:

[0039] Safe Space™ 216 is an "intelligent hearing" application that detects and outputs a response based on the identification of a recognized sound. The concept of recognizing sounds is described as Sonic Signature Detection. Safe Space™ 216 also includes a user defined hierarchy and can provide responses based on the hierarchy.

[0040] A brief example is provided of how Safe Space™ 216 manages the acoustic space corresponding to the interior of an automobile. As is well known, automobiles are designed to have a quiet interior space. Sound insulation is deliberately placed around the car interior to attenuate sounds coming from the car exterior. Furthermore, automobile drivers often listen to high volume music while driving. This combination makes it difficult to hear sounds such as emergency vehicles. People often get seriously injured because they hear the emergency vehicle when it is in close proximity to the vehicle and panic causing an accident. All descriptions below describing personalized sound management applications assume that the device using the application has incorporated within it the hardware and software required to run the application and perform the application function.

[0041] In this example, Safe Space™ 216 detects emergency vehicle horns when they are a substantial distance away. In at least one exemplary embodiment, a microphone is exterior to the car and can pick up sounds in the ambient or car exterior. Sonic signatures related to emergency vehicles such as fire

trucks, ambulances, and police cars are stored in the system. Safe Space™ 216 analyses sounds from the microphone. A response is provided if one of the stored sonic signatures is detected. In at least one exemplary embodiment, upon detecting a fire truck siren, Safe Space™ 216 can initiate playing the identified signal through the car stereo system for the car driver to hear and respond to. In at least one exemplary embodiment, Safe Space™ 216 can calculate the direction, distance, and street (through GPS) of the approaching emergency vehicle. The information is then provided visually or vocally to the driver. For example, the car stereo automatically turns down the music and states through the speaker system that an ambulance is coming east bound on 3.sup.rd Street or an ambulance is approaching from the right.

[0042] EarGuard™ 218 is a personalized sound management application that improves listening quality and safety. Hearing disorders are increasing at a very high rate. Many people live in an environment such as a noisy urban environment or an industrial manufacturing area where sound pressure levels are consistently high. Moreover, people themselves are subjecting themselves to loud and sustained sounds. Examples of extreme or sustained sound exposure are portable media players using earbuds, sport shooting, and rock concerts. The trend is a growing population of people that have or will have hearing problems.

[0043] The ear is a very sensitive instrument having a large dynamic range that allows us to hear sounds ranging from a whisper to a shout. Subjecting the ear to man-made cacophony is now known to metabolically exhaust the highly specialized sensory cells in the inner ear, causing them to die and be replaced by scar tissue in the hearing organ. Ear Guard™ 218 is an application that monitors the sound exposure of the user and protects the ear from damage.

[0044] Briefly, Earguard™ 218 improves listening quality and safety by employing a personal hearing zone that measures sound levels within the ear and in one exemplary embodiment an intelligent level control to protect against over amplification. Earguard™ 218 includes a sound pressure level (SPL) dose management system. The SPL dose management system takes into account both transient and long term sound. In at least one exemplary embodiment, when sound at the user's eardrum is above a certain threshold known to begin the process of metabolically exhausting the ear, the metric known as SPL Dose increases. The SPL dose decreases when sound is below that threshold according to a recovery function that mirrors the recovery of the ear from excessive sound exposure. In at least one exemplary embodiment, Earguard™ 218 will indicate to the user damage to the ear is likely if the sound exposure is continued. In this scenario, the user has the control to take the appropriate action to protect his/her ears.

[0045] In at least one exemplary embodiment, an intelligent level adjustment system is a personalized sound management application that automatically estimates the preferred listening level of an audio content signal (e.g. speech or music audio from a device) depending on an analysis of the level, other acoustic features of the ambient environment, and an analysis of the audio content signal. Thus, a human-machine relationship is nurtured with this bi-directional control flow from human to the intelligent level adjustment system and from intelligent level adjustment system to the human user.

[0046] A substantial benefit of an application such as Earguard™ 218 is the protection of the ear from

damage. Furthermore, this safeguard if propagated will prevent hearing loss thereby reducing the financial burden on the economy in years to come for the individual, business, and government for hearing loss remedies. Thus, using Earguard™ 218 could not only stem the growth in hearing related problems but greatly reduce it over the years to come.

[0047] Quiet Call™ 220 is an application that allows user 202 having heard a voice message to respond to the remote caller through a non-verbal means. An example of a non-verbal response is a key-pad entered text message. The entered text message is converted to a speech audio message and is sent to the remote caller. The caller then receives the speech audio message.

[0048] An example of the utility of Quiet Call™ 220 is illustrated when user 202 is in an important business meeting but is required to provide input to another remote meeting. User 202 receives the voice message sent from someone in the remote meeting. User 202 responds by entering the response through his/her phone keypad; the text is converted to voice and sent to the person at the remote meeting who sent the original message. User 202 does not have to interrupt the meeting to listen to the voice message nor to reply. Thus, the meeting attended by user 202 can move forward with little or no loss of momentum that would occur if the meeting was disrupted by the call.

[0049] Hearable™ 222 is a speech enhancement application that improves voice communication. For example, Hearable™ 222 can be used with an earpiece device having at least two microphones, an ambient sound microphone for receiving sounds in the ambient environment and an ear canal microphone for receiving sounds in the ear canal. A common use for the earpiece would be with a communication device such as a cell phone or other phone system. The earpiece using the Hearable™ 222 application would normally use the ambient sound microphone for receiving and transmitting the user's spoken voice. In this mode, the user's voice will be natural sounding and easily recognizable on the receiving end.

[0050] In a noisy environment it can be difficult to carry on a telephone conversation. The ambient sound microphone will pick up the voice and the noise in the ambient environment. The earpiece will switch to the ear canal microphone when the ambient sound microphone detects a high background noise level. The user's voice is readily picked up by the ear canal microphone but the noise in the ambient environment is substantially reduced. Switching to the ear canal microphone allows the receiving party to clearly hear the user's voice. The problem with using the ear canal microphone is that user's voice received in the ear canal sounds different because of frequency roll off in the upper spectrum of the voice range. Although the user can be heard clearly, the user's voice may not sound right to the receiving end.

[0051] Hearable™ 222 is a personalized sound management application that improves the sound quality of the user's voice. Hearable™ 222 uses a combination of the sound received by the ambient sound microphone and the ear canal microphone to create a more natural sounding voice. The combination of the two signals is a function of the background noise level. Explaining further the signal from the ambient sound microphone is used less as the background noise level increases. Hearable™ 222 allows a user to have a conversation in a noisy environment while providing a high quality voice signal that is intelligible and recognizable at the receiving end.

[0052] Always On-Recording™ 224 is a personalized sound application that acts as its name implies. A device using the Always On-Recording™ 224 application is recording the most current audio information for user recall. The recording is stored in a buffer that will allow the user to immediately access the audio information. The buffer will have a finite amount of storage. The recording time will be a function of the buffer or memory available for the Always On-Recording™ 224 application.

[0053] Always On-Recording™ 224 provides utility for short durations of recording. For example, in an application where a user is receiving information such as a phone number or driving directions. The user knows that the device employing Always On-Recording™ 224 has the stored acoustic information and can immediately listen-again to the buffer contents thereby repeating the phone number or driving directions. Similarly, if the user was discussing a contract term and wanted to know exactly what the person said, the user could immediately re-listen to make sure that what they thought they heard is exactly what they heard.

[0054] Earbliss™ 226 is a personalized sound management application to provide acoustic isolation from someone who snores while allowing other sounds to be heard. A large percentage of the population suffers though sleepless nights because of the sounds generated by people with sleep apnea and more generally due to loud snoring. Moreover, sleep deprivation can have serious consequences related to health and disposition.

[0055] Earbliss™ 226 is a sleep zone technology that utilizes sonic signature detection to insulate sleep partners against intrusion from snoring while still providing awareness of priority sounds. In this example, a sonic signature is acoustic information related to the sounds of a person snoring. In at least one exemplary embodiment, Earbliss™ 226 is an application that is used in conjunction earpieces.

[0056] Sealing the ear canal with ear plugs will attenuate the snoring but also blocks out all sounds. Hearing is one of the most vital senses that we have. Under normal conditions we cannot turn off our hearing which allows us to wake in a critical situation. Ear plugs will block out sounds of consequence. For example, the user may not hear their baby crying or a child leaving their room.

[0057] Earbliss™ 226 enables a user to attenuate snoring while hearing other sounds around them. The earpieces fit in the ear canal of the user to seal or partially seal the ear canal. The earpieces have an ambient sound microphone for receiving acoustic information from the ambient environment and an ear canal receiver for providing sound to the user's ear canal. As mentioned previously, the earpieces will have a sonic signature related to snorer's snoring stored in memory in the device.

[0058] The ambient sound microphone picks up all sounds in the ambient environment including the snorer's snoring. The earpiece processes all the acoustic information coming from the ambient sound microphone and looks for signals similar to stored sonic signatures. Pattern recognition approaches are applied based on the known sonic signatures to detect the snoring sounds from their corresponding sonic signatures. More specifically, sonic signatures can then be compared to learned models to identify a corresponding snoring sound. Once identified the snoring sound is suppressed and not output to the ear canal by the ear canal receiver. Thus, the wearer of the earpieces does not hear the snoring.

[0059] Conversely, sounds in the ambient environment that are not recognized through the processing of acoustic information by the earpieces can be passed transparently to the ear canal receiver for reproduction within the ear canal. In this mode, the sound produced in the ear canal sufficiently matches the ambient sound outside the ear canal, thereby providing a "transparency" effect other than the suppressed sonic signature (snoring). The earpieces can also enhance sound. For example, the earpieces having sonic signatures related to a fire truck siren or a baby crying can detect either signal and then amplify the signal (fire truck siren, baby crying) so as to make the wearer of the earpieces aware of the signal's detection. Thus, Earbliss™ 226 modifies the acoustic environment of the user to eliminate what does not need to be heard while allowing the user to be aware of other sounds in a normal context.

[0060] Ear Mail™ 228 is a personalized sound management application for converting text to voice. In particular, Ear Mail™ 228 provides great utility to users of email and text messaging although not limited to these examples. Email and text messaging is becoming a very popular form of communication among a large portion of the population. It is not always convenient or in some cases not prudent to review written text depending on the situation.

[0061] Ear Mail™ 228 converts the text of a message or email to a speech audio message using a text to speech algorithm. The converted speech audio message is replayed through a loudspeaker coupled to a device using the Ear Mail™ 228 application. For example, Ear Mail™ 228 used in a smart phone coupled to the stereo system of a car through a blue tooth connection could playback the text or emails through the car speakers. The user could hear their messages while driving safely down the road.

[0062] Ear Sticky™ 230 is a personalized sound management application for recording information that can be saved and used for future use. In at least one exemplary embodiment, Ear Sticky™ 230 is a mobile communication application that can be used in a device such as a cell phone or an earpiece that is operably coupled to other devices. Ear Sticky™ 230 can record communication through a device or sounds in the ambient environment. Sounds in the ambient environment are recorded by an ambient sound microphone.

[0063] In a first example of an Ear Sticky™ 230 application, a conversation between a husband and wife occurs and a list of items to pick up at several stores is disclosed. The user of Ear Sticky™ 230 does not need to write down or remember this list. The conversation is being recorded and stored in a buffer. The user activates Ear Sticky™ 230 to store the recorded conversation to be reviewed at a later time. Thus, the user of Ear Sticky™ 230 could recall and listen to the list on the way home to ensure that the right items are picked up at the appropriate store.

[0064] In a second example of an Ear Sticky™ 230 application, a device records with an ambient sound microphone. For example, the user of Ear Sticky™ 230 comes up with a great concept to solve a problem he or she has been working on for some time. The user can enable Ear Sticky™ 230 and uses the ambient sound microphone to record his or her voice to convey the idea, concept, or thoughts and store it for review at a later time. In general, Ear Sticky™ 230 provides utility and convenience in storing and recalling sounds in one's acoustic space.

[0065] Hear & Tell™ 232 is a personalized sound management application for recording a sound, training a Gaussian mixture model to learn features of the sound and then storing the Gaussian mixture model in memory of the device. A user of Hear & Tell™ 232 can record the sound or provide a sound, called herein as a sonic signature. The device is operably coupled to at least one microphone to compare sounds received from the microphone against the stored sonic signature. The device can perform an operation that modifies the user's acoustic space 204 once a sound is identified as being similar to a sonic signature. Examples of several operations the device can perform are passing the detected signal through to the user, boosting the sound such that the user is made aware of the detected sound, rejecting the sound so the user does not hear the detected sound, attenuating the detected sound, and replacing the detected sound with an alternate sound to name.

[0066] The Hear & Tell™ 232 application will store a Gaussian mixture model (GMM) for every sonic signature that it has been trained to recognize. Each GMM is completely specified by a mixture of mean vectors, a mixture of covariance matrices, and a mixture of weights.

[0067] An example of a warning sound (e.g. siren of emergency vehicle) will be used to further illustrate the Hear & Tell™ 232 learning application. Each GMM provides a model for the distribution of the feature statistics for each warning sound in a multi-dimensional space. Upon presentation of a new feature vector, the likelihood of the presence of each warning sound can be calculated. In at least one exemplary embodiment, each warning sound's GMM is evaluated relative to its anti-model, and a score related to the likelihood of that warning sound is computed in order to determine if a sound is detected. A threshold can be applied directly to this score to decide whether the warning sound is present or absent. Similarly, a sequence of scores can be relayed and used in a more complex rule set to determine absence or presence of the sound. Thus, Hear & Tell™ 232 allows a user to store, model, and train a device for sonic signatures under user control and selection thereby allowing a user to modify their acoustic space through the detection of a sound and response by the device.

[0068] In general, personalized sound management applications are customizable point solutions that allow user 202 to handle a variety of tasks associated with managing acoustic space 204. The selection of personalized sound management applications will depend on the person, the device, and the acoustic space being managed. Examples of devices that can use personalized sound management applications with the appropriate hardware and operating system, but not limited to these examples, are earpieces, media devices, and vehicles.

[0069] As described hereinabove, a suite of personalized sound management applications 206 are provided. The user 202 can select the desired applications and have control over their acoustic space 204. The selected applications are stored in ROM 212. Under user control, user 202 selects the parameters of personalized sound management applications 206 using operating system 208 for implementing applications 206 in device 210 for managing acoustic space 204 based on the individual's needs and wants.

[0070] FIG. 3 is a diagram illustrating a module 300 for implementing personalized sound management in accordance with at least one exemplary embodiment. Module 300 comprises an H-Chip™ 302, a ROM 304, an operating system 306, and user selected personalized sound management applications 308. In at

least one exemplary embodiment, module 300 comprises the H-Chip™ 302 with ROM 304 built into H-Chip™ 302. Both operating system 306 and user selected personalized sound management applications 308 are stored on ROM 304. Alternately, H-Chip™ 302 and ROM 304 can be separate chips allowing for a larger block of memory.

[0071] H-Chip™ 302 is a microprocessor, DSP, or logic circuit that implements personalized sound management applications 308. H-Chip™ 302 is optimized for low power dissipation while running user selected personalized sound management applications 308. In at least one exemplary embodiment, H-Chip™ 302 can be a dedicated engine for running applications 308.

[0072] Module 300 comprises an application engine (firmware) and a dedicated processor that simplifies the integration of user selected and user controlled personalized sound management applications 308 into a device. In an exemplary embodiment, module 300 is integrated into each device in devices 314. Devices 314 typically, although not always have at least one microphone and at least one speaker. User 310 in conjunction with module 300 can personalize how each device manages sound to the user.

[0073] An additional aspect of module 300 is that a third party manufacturer merely builds the acoustic portion of their device around module 300. In an exemplary embodiment, a manufacturer does not and cannot modify operating system 306 and user selected personalized sound management applications 308. Thus, the manufacturer saves time, effort, and money through the seamless integration module 300 into a single hardware solution for maximum transportability into 3.sup.rd party form factors.

[0074] FIG. 4 is a diagram illustrating a process of providing personalized sound management in accordance with at least one exemplary embodiment. In general, the technology related to providing personalized sound management comprises hardware components and software. In at least one exemplary embodiment, the hardware 402 comprises transducers 404 and H-Chip™ 406.

[0075] Transducers 404 are speaker and microphones for respectively providing and receiving sound in an acoustic space. Depending on the application, the size and form factor of transducers 404 may be a critical design parameter. Transducers 404 can be a speaker for providing high fidelity sound to enhance a user's experience. Similarly, transducers 404 can be a microphone for receiving acoustic information and in some cases can pick up sounds inaudible to the user.

[0076] H-Chip™ 406 can be a microprocessor, digital signal processor, logic circuit, or applications specific integrated circuit for use implementing the software applications listed hereinabove and other programs related to personalized sound management. H-Chip™ 406 includes operating system 408 specific for managing the device as it relates to personalized sound management and allows user control to adjust the parameters of each application for their specific needs.

[0077] Hardware 402 can be integrated into existing devices or next generation devices for adding utility, providing safety, and allowing personalization of a user's acoustic space. Providing hardware 402 enables device manufacturers 410 to rapidly integrate the technology into their products. This will quicken the adoption cycle of the technology for the benefit of the general public in health and safety, and for the individual through personalization. Hardware 402 can be a circuit board to which

transducers 404 and H-Chip™ 406 are operatively attached.

[0078] Device manufacturers 410 can provide their own hardware 414 or use transducers 404 and H-Chip™ 406. The operating system and application software is incorporated 412 and stored in read only memory (ROM). In at least one exemplary embodiment, operating system 408 and the application software can be stored in ROM. Using read only memory for storage prevents device manufacturers 410 or consumers from tampering with the software code thereby maintaining the integrity of the system and how it performs for the user. Note herein when referring to ROM storage, at least one exemplary embodiment can include RAM or other read/write storage methods.

[0079] People in the consumer product industry have stated that any new product and even old products in the hands of consumers may produce liability issues that companies associated with the product have to deal with. This often delays an introduction of a product or adds substantial cost to a product launch. In at least one exemplary embodiment, device manufacturers 410 provide their product for certification 416. In at least one exemplary embodiment, certification 416 is performed by an independent company from device manufacturers 410. Certification 416 is a process whereby a new product is exhaustively tested to ensure the device will perform to specification under a variety of conditions and tolerances. In particular, certification 416 tests the operating system and the device performance in implementing the applications related to personal sound management. Should the device not pass certification testing, it can be repaired or redesigned. The repaired or redesigned device can then undergo certification 416 to determine if it meets the test specifications.

[0080] In at least one exemplary embodiment, an insurance policy 418 may be provided covering the device manufacturers 410 using operating system 408, hardware 402, and the application software. This provides substantial benefits in that the device manufacturers 410 can deploy the technology with less risk. Moreover, each company providing devices will know that the personalized sound management system is similar or identical to others in the market. As mentioned hereinabove, the operating system and application software can be stored in ROM ensuring it cannot be rewritten. Additional measures can be taken to determine if the software is being used out of context and shutting down the operation of the device.

[0081] Once tested and certified, device manufacturers 410 can manufacture, market, and sell devices with personalized sound management 424 and be covered from a liability perspective under provisions in the insurance policy 418. A consumer purchases 420 a certified device and may also be covered under the same insurance policy 418 or may have a second policy directed to the consumer. Consumer protected 422 under insurance policy 418 and having certification 416 will instill consumer confidence to the quality and reliability of the device. This will be discussed in greater detail hereinbelow. Thus, this process is very efficient in creating manufacturer adoption of the personalized sound management technology while protecting both device manufacturers 410 and the consumer.

[0082] FIG. 5 is a diagram illustrating application testing in accordance with at least one exemplary embodiment. Device manufacturers 502 develop, prototype, and manufacture products 506. Products 506 include hardware and operating system 504 for running and performing Personalized Sound Management (PSM™) applications 508.

[0083] In at least one exemplary embodiment, hardware and operating system 504 is capable of running all personalized sound management applications 508. Products 506 may or may not use all of personalized sound management applications 508. Similarly, an end user or customer of products 506 may or may not use all of personalized sound management applications 508.

[0084] Hardware and operating system 504 of products 506 are loaded with all personalized sound management applications 508. The hardware and operating system 504 are exercised in testing 510 to ensure that it can run personalized sound management applications 508 per specification. Product 506 may or may not be tested for all personalized sound management applications 508 depending on the capability of the device. Products 506 will be tested 510 for the implementation of all personalized sound management applications it is capable of running to meet user specifications for the product (i.e. product certification 512). In at least one exemplary embodiment, should a product run only a subset of personalized sound management applications, the product can be designed and tested to lock out the running of applications not specified for the particular product (even though hardware and operating system 504 can run personalized sound management applications 508).

[0085] A scale of economy for developing and manufacturing hardware and operating system 504 is achieved by having a number of different device manufacturers 502 standardize on the engine (hardware and operating system 504) for implementing personalized sound management. Consistency of product, ability to expand scope of product, and lowering cost through volume manufacturing are all achieved by this methodology. In the example where liability insurance may be purchased by the product certification company (or other company) testing 510 all personalized sound management applications 508 in products 506 ensures that hardware and operating system 504 perform per specification independent of the product it is placed in. Furthermore, having little or no variation on the operation of hardware and operating system 504 and personalized sound management applications 508 minimizes risk. An added benefit is that a large statistical database of diversified products is generated that will allow improvements to managing personalized sound management devices to improved device performance as well as lower liability risk. The consumer benefits by having a highly exercised and stable product that improves their health, safety, and quality of life.

[0086] FIG. 6 is a diagram illustrating testing of a personalized sound management product 602 in accordance with at least one exemplary embodiment. In at least one exemplary embodiment, testing of all products incorporating personalized sound management applications are tested by a single entity. Personalized sound management product 602 can be developed and manufactured by a number of different companies. As previously mentioned, this allows the single entity, should it so desire to purchase a liability insurance policy that covers the manufacturers making personalized sound management product 602 and consumers using personalized sound management product 602.

[0087] Personalized sound management product 602 is tested by certification laboratory 604. In at least one exemplary embodiment, certification laboratory 604 is independent from the different manufacturers that develop personalized sound management product 602 thereby providing unbiased testing. Testing 614 of hardware 606, software 608, and manufacturing 610 are performed on personalized sound management product 602. Testing 614 is related to operation and performance of

personalized sound management applications. Hardware testing includes the processor, transducers, and other elements, which implement the personalized sound management of the device. Testing of software 608 includes the operating system and personalized sound management applications. Testing 614 includes reliability 616 and quality 618 testing. Testing 614 can also include human testing because in some embodiments, personalized sound management product may be worn in an orifice of the human body.

[0088] Upon passing testing 614 the personalized sound management product 602 is certified by certification laboratory 604. A product certified 620 can include some form of warranty, indemnification, and liability insurance for the device. Problems or issues with personalized sound management product 602 are reported to the company should it fail for any reasons. Corrective action can be taken and the device retested to determine if the issue has been resolved.

[0089] In at least one exemplary embodiment, certification laboratory 604 or its parent company will brand the product by placing a trademark or company logo on the approved personalized sound management product 602. The trademark or logo represents the incorporation of its personalized sound management technology in the product (hardware, software, and intellectual property) as well as the certification indicating the quality and reliability of the device.

[0090] FIG. 7a is a diagram illustrating a consumer purchase process for a personalized sound management product in accordance with at least one exemplary embodiment. A consumer reviews a number of different devices such as earpieces in a display area. The display area allows the consumer to look, feel, and touch the hardware. The consumer focuses on an earpiece 701 in the display area and wants to learn more about the device.

[0091] A sales person can demonstrate earpiece 701 allowing the consumer to test the functions and performance of the device. A demonstration room or test environment can be used to illustrate performance differentiation when compared to other earpieces. In at least one exemplary embodiment, earpiece 701 is an in-ear device that has a removable and disposable sealing section. This allows consumer testing using earpiece 701 while maintaining a sanitary condition by disposing of the sealing section after each test session.

[0092] The consumer indicates to the sales person that he or she would like to purchase earpiece 701. The sales person accesses a computer that is coupled to server system for completing a purchase transaction. The consumer provides registration information that is entered into a database in a step 703. The consumer or the sales person can enter the information into the database. The registration information 705 comprises information such as personal, financial, business, and preference. Personal information relates to data such as age, sex, home address, telephone number, email address, which identifies the consumer. Financial information relates the consumer's ability to pay such as credit history, credit card for billing, job information, and banking institutions. Business information relates to a business purchase of the component. Similar to above, business identification and business financial information would be entered with the purchase. Preferences relate to things such as how the consumer wants to be contacted (email, phone), do they want to be made aware of upgrades for their device, and other products they might have interest in. Other information or user experiences could also

be collected during the registration process.

[0093] The unit selected by the consumer is then entered into the system in a step 707. In at least one exemplary embodiment, the system identifies the unit and provides a listing of the hardware, software, and subsystems that work with the earpiece 701. The consumer can look at the options available in a step 709 and select among these options. In at least one exemplary embodiment, the selection process can be facilitated by topic, occupation, or other characteristics by providing a list of typical user features applied to earpiece 701. For example, the consumer is purchasing earpiece 701 for a multimedia player. Earpiece 701 provides a substantial improvement in noise attenuation, musical clarity, definition, and dynamic range. The consumer is interested in the safety aspect (and selects this topic on the kiosk) provided by earpiece 701 in alerting the user to an emergency event. For example, earpiece 701 can detect and make the user aware of sirens, emergency sounds, alarms, etc. that would be difficult to hear when listening to music. The consumer by looking at the "safety" topic can select appropriate applications software and sonic signatures to alert the user of a safety situation for their specific device. The consumer can continue to look through other topics and select hardware and software that supplements and implements the desired user experience.

[0094] As mentioned previously, the options can be provided in a step 709 in many different formats. Another format of providing options is by occupation. For example, a dentist may be provided a package to outfit the entire office and patients. The earpieces substantially reduce noise related to drilling which are both a psychological barrier for dentistry and a hearing loss mechanism for people working in a dentist office. The occupation package can provide wireless communication between devices allowing dentists, assistants, and patients to talk or listen to music even in a noisy environment. Replaceable in-ear sealing sections would allow the staff to maintain sanitary conditions with each new patient. After reviewing the consumer selects the appropriate hardware and software to personalize their sound space (on the kiosk or computer) in a step 711.

[0095] In at least one exemplary embodiment, the selected application software provided to earpiece 701 is a subscription whereby the consumer pays a monthly fee to use the applications. Alternately, the selected application software can be a license or purchase associated with the earpiece 701 whereby a one time fee is paid. The consumer can discuss the terms of the agreement with the sales person or review the contract on the computer. If the consumer approves, the contract is signed or accepted and an initial payment is made for the hardware and software in a step 713. The consumer can have the subscription payment automatically billed to a credit card or other automatic payment methodology to ensure uninterrupted service.

[0096] The purchased earpiece is then connected via a wired or wireless connection to download the selected software and enabled in a step 715. A wired connection 717 such as a mini USB cable is shown in the illustration. Wired connection 717 connects to a server system having the appropriate software. In at least one exemplary embodiment, the selections from step 711 can be stored in memory and used to direct the downloading of software to the device, verify the software can be used in the device, and that the appropriate software has been loaded.

[0097] FIG. 7b is a diagram illustrating a consumer purchase process for a personalized sound

management product in accordance with at least one exemplary embodiment. A consumer 702 can purchase personalized sound management product hardware 704 and personalized sound management application software 706. Software 706 works with hardware 704 to personalize and manage sound for consumer 702 for providing utility, safety, and health benefits.

[0098] Hardware 704 can range from a complete product such as a communication device, sound device, or earpiece to ancillary components for enhancing the look of the device or adding further hardware features. In the event that consumer 702 purchases hardware 704, customer information 708 is required to continue with purchase. In general, customer name, address, personal information and financial information are typically required to purchase hardware 704. Other information can also be collected through a questionnaire or some form of incentive. Should customer information 708 exist and consumer 702 is in good stead then generating customer information 708 is not required and the order process continues. Any issue that occurs in checking customer information 708 will be brought up for rectification to consumer 702. The hardware 710 that consumer 702 selected is verified and place order 712 is performed.

[0099] Personalized sound management application software 706 is a suite of applications that consumer 702 selects for their specific needs. Once customer information 708 has been established, consumer 708 is provided subscription information 714. Use of personalized sound management application software 706 is based on a periodic fee or subscription (for example, a monthly fee). Subscription information 714 informs consumer 702 as to the periodic subscription fees associated with the software 706, terms of use, bundled packages of applications, and other pertinent information to complete the decision process. Should consumer 702 approve of the terms, a contract 716 is provided to the customer to accept or decline. A bill 718 is generated when an order is placed either through purchasing hardware 710, accepting a subscription to personalized sound management application software 706, or both. In at least one exemplary embodiment, the purchase process is web based allowing consumer 702 to purchase from a client system via a communication path such as the internet or communication network.

[0100] FIG. 8 is a diagram illustrating registering a new device that includes personalized sound management applications in accordance with at least one exemplary embodiment. A new device or unit is received by a customer 802. The manufacturer has loaded the preordered personalized sound management applications chosen by customer 802 into the unit. Typically, customer 802 is going to want to try the unit shortly after receiving it (e.g., the unit is operational for a predetermined period of time 804, for example 1 week). In at least one exemplary embodiment, customer 802 can try the unit for a predetermined time period 804. In general, the time period 804 (for example, 15-30 minutes) is long enough for customer 802 to get a feel for the unit but is not a useful length of time. Furthermore, the unit will notify customer 802 through a transducer on the unit that they must register the device and that the unit will not be usable after the predetermined time period. Note that in at least one exemplary embodiment the customer would have the opportunity to demo certain features (806) not purchased.

[0101] In at least one exemplary embodiment, a website 810 is a portal to register the unit purchased by customer 802. Website 810 can have a web page devoted to registering new devices to simplify the process. Customer 802 may log in (if an account already exists) or provide personal information related

to the registering process. A database 812 is accessed that includes both customer and order information in one or more databases. In at least one exemplary embodiment, the unit is in communication with the device manufacturer server 814 through website 810 or coupled thereto through a customer system.

[0102] Verification 816 verifies that all information is correct and that there are no pending issues 818 with a corrective action identified. The verification process 816 can checks things like device serial number, features and applications ordered by customer 802, personal/financial information. Once verification 816 is complete, it is determined at step 817 whether to register the device. Step 817 proceeds to step 820 if there are no pending issues 818. Registration 820 registers the unit to the owner after verification 816 is complete.

[0103] FIG. 9 is a diagram illustrating enabling a device in accordance with at least one exemplary embodiment. In general, registration 902 links customer information to a device as disclosed in FIG. 8. The registration process includes a comprehensive legally-compliant, state-specific Informed Consent system for collecting and storing the Informed Consent of users.

[0104] In at least one exemplary embodiment, the device will have at least one microphone. In the process of registration 902 the customer voice is recorded by the device. The customer voice can be stored on the device and to the customer database. The customer voice can be used for a number of applications including voice verification to use the device thereby acting as a deterrent for others to use or steal the device.

[0105] As mentioned in FIG. 8 the unit or device can be enabled for a predetermined time period. Prior to enabling the new device, a contract 904 is provided for customer review. The contract outlines the terms of use for using the device and that the product had been certified through a rigorous testing process. In at least one exemplary embodiment, liability coverage may be provided to the customer if the contract is accepted. Having the liability coverage in conjunction with the certification process is powerful in building consumer confidence about personalized sound management.

[0106] At step 905, it is determined whether the contract is accepted. The new device is not enabled 906 if the customer does not accept contract 904. The new device is enabled 908 if the customer accepts contract 904. In at least one exemplary embodiment, the registration and enabling of the unit is through a client system coupled to the business server. Thus, the unit is enabled remotely.

[0107] FIG. 10 is a diagram illustrating updating a unit or device in accordance with at least one exemplary embodiment. The device is enabled for use. In general, instructions or an operational video may be provided for a user to get acquainted with the device. Even with this help, a device may have so many features and applications that the user could get overwhelmed or frustrated trying to understand and operate the unit. In some cases, a user may not want even try to learn how to operate all the features and initially start out with the application that provides significant utility to the user. The user may never address learning how to use all the capabilities of the device. Moreover, misuse of a device can lead to dangerous operating conditions that could cause a serious or harmful situation.

[0108] Acclimation 1002 is a process to ensure that a user of a device appreciates the entire feature set but more particularly, enables the features in a way that ensures proper usage in a safe and structured manner. In at least one exemplary embodiment, acclimation 1002 comprises two components, a learning module and a reminder module. The learning module is a system that "teaches" the user of the device the functional components of the device. In at least one exemplary embodiment, the learning module may incorporate a learning system that evaluates the user in the domain of a particular competency before certain functions or features of the device are enabled. For example, features of the device are enabled sequentially, either automatically after the user has utilized a particular function for a predetermined time (thereby showing competency in using the feature) or following a user competency evaluation, which can be invoked either manually (when the user is confident in using the feature) or automatically.

[0109] The reminder module may or may not be used in acclimation 1002. The reminder serves to remind the user of the correct usage of features and also serves to remind the user of other features on the device which the user may not have used in a given time period. In at least one exemplary embodiment an evaluation is invoked based on a predetermined time period. The evaluation may be for a single function or feature or a combination of different functions. The predetermined time period before evaluating can vary depending on the function. Alternatively, the evaluation may be invoked based on a particular recent event, such as a software upgrade, or a detected change in user usage. If the user passes the evaluation, the evaluation may be repeated, the relevant learning mode for the corresponding function to which the users competency was evaluated may be repeated, or the corresponding function to which the users competency was evaluated may be disabled. Thus, acclimation 1002 allows a user to learn the device at a friendly pace, ensures that the user does know how to use a feature, and protects the user from using the device in a manner that could be harmful.

[0110] The device can be coupled 1004 from a client system to the business server to update software on the device or purchase new hardware and software. As disclosed hereinabove, the user information is checked to determine if there are any issues that need to be rectified (e.g. missed payment, user does not match to device, etc. . . .). In the event that the issues cannot be rectified the device is disabled in a step 1006. In at least one exemplary embodiment, the user information, device, and personalized sound management applications correlate. The server checks for updates 1008 for the device and downloads updates to the device. The user goes through a similar process as described in FIGS. 7A-7B for a hardware or software purchase 1010 including agreeing to a contract to subscribe and contract to terms of use as described in FIG. 9.

[0111] FIG. 11 is a diagram illustrating a device for implementing personalized sound management in accordance with at least one exemplary embodiment. The device is generally indicated as an earpiece that partially seals or seals a user's ear canal 1124 and is constructed and operates in accordance with at least one exemplary embodiment of the invention. As illustrated, the earpiece comprises an electronic housing unit 1100 and a sealing unit 1108. The earpiece depicts an electro-acoustical assembly for an in-the-ear acoustic assembly, as it would typically be placed in an ear canal 1124 of a user 1130. The earpiece can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, partial-fit device, or any other suitable earpiece type. The earpiece can partially or fully occlude ear canal 1124, and is suitable for use with users having healthy or abnormal auditory functioning.

[0112] The earpiece includes an Ambient Sound Microphone (ASM) 1120 to capture ambient sound, an Ear Canal Receiver (ECR) 1114 to deliver audio to an ear canal 1124, and an Ear Canal Microphone (ECM) 1106 to capture and assess a sound exposure level within the ear canal 1124. The earpiece can partially or fully occlude the ear canal 1124 to provide various degrees of acoustic isolation. In at least one exemplary embodiment, the assembly is designed to be inserted into the user's ear canal 1124, and to form an acoustic seal with the walls of the ear canal 1124 at a location between the entrance to the ear canal 1124 and the tympanic membrane 1126 (or ear drum). In general, such a seal is typically achieved by means of a soft and compliant housing of sealing unit 1108.

[0113] Sealing unit 1108 is an acoustic barrier having a first side corresponding to ear canal 1124 and a second side corresponding to the ambient environment. In at least one exemplary embodiment, sealing unit 1108 includes an ear canal microphone tube 1112 and an ear canal receiver tube 1110. Sealing unit 1108 creates a closed cavity of approximately 5 cc between the first side of sealing unit 1108 and the tympanic membrane 1126 in ear canal 1124. As a result of this sealing, the ECR (speaker) 1114 is able to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum 1126 resulting from the sound field at the entrance to the ear canal 1124. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0114] In at least one exemplary embodiment and in broader context, the second side of sealing unit 1108 corresponds to the earpiece, electronic housing unit 1100, and ambient sound microphone 1120 that is exposed to the ambient environment. Ambient sound microphone 1120 receives ambient sound from the ambient environment around the user.

[0115] Electronic housing unit 1100 houses system components such as a microprocessor 1116, memory 1104, battery 1102, ECM 1106, ASM 1120, ECR 1114, and user interface 1122. Microprocessor 1116 (or processor 1116) can be a logic circuit, a digital signal processor, controller, or the like for performing calculations and operations for the earpiece. Microprocessor 1116 is operatively coupled to memory 1104, ECM 1106, ASM 1120, ECR 1114, and user interface 1122. A wire 1118 provides an external connection to the earpiece. Battery 1102 powers the circuits and transducers of the earpiece. Battery 1102 can be a rechargeable or replaceable battery.

[0116] In at least one exemplary embodiment, electronic housing unit 1100 is adjacent to sealing unit 1108. Openings in electronic housing unit 1100 receive ECM tube 1112 and ECR tube 1110 to respectively couple to ECM 1106 and ECR 1114. ECR tube 1110 and ECM tube 1112 acoustically couple signals to and from ear canal 1124. For example, ECR 1114 outputs an acoustic signal through ECR tube 1110 and into ear canal 1124 where it is received by the tympanic membrane 1126 of the user of the earpiece. Conversely, ECM 1106 receives an acoustic signal present in ear canal 1124 though ECM tube 1112. All transducers shown can receive or transmit audio signals to a processor 1116 that undertakes audio signal processing and provides a transceiver for audio via the wired (wire 1118) or a wireless communication path.

[0117] The earpiece can actively monitor a sound pressure level both inside and outside an ear canal

1124 and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

[0118] The earpiece can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 1124 using ECR 1114 and ECM 1106, as well as an Outer Ear Canal Transfer function (OETF) using ASM 1120. For instance, the ECR 1114 can deliver an impulse within the ear canal 1124 and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal 1124. The earpiece can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece to safely administer and monitor sound exposure to the ear.

[0119] In at least one exemplary embodiment, the earpiece has a number of sonic signatures stored in memory. ASM 1120 is providing acoustic information from the ambient environment to processor 1116. Processor 1116 analyses the acoustic information for a sound similar to the sonic signature. Once identified, the earpiece will provide a response to the sound based on the application. In a first exemplary embodiment, the earpiece will reduce music or telephone call (or the dominant source of sound being provided by the earpiece) and amplify the identified signal (ambulance or police car) thereby notifying the user of the approaching vehicle. In a second exemplary embodiment, the earpiece will tell the user (through a synthesized voice) that an ambulance or police car is approaching including the direction of the vehicle. The earpiece can also provide the identified signal with the voice warning. Other variations are possible.

[0120] Conversely, the earpiece can perform the opposite operation. The earpiece can identify a signal similar to the sonic signature and then attenuate it before providing it through ECR 1114. For example, the user of the earpiece is a gun enthusiast. The user downloads a sonic signature related to a gun shot. The earpiece upon identifying the sound of the gun shot would attenuate the portion of the acoustic information provided by ASM 1120 similar to the sonic signature of the gun shot while allowing other signals to come through. Thus, the user could engage in a conversation at the gun range with the gun shot sounds attenuated while passing the conversation through the earpiece thereby protecting his ear from the loud sounds in this environment and being able to hear the conversation with more clarity.

[0121] In at least one exemplary embodiment, the earpiece can manually or automatically record, measure sound pressure levels, attach metadata (including time stamp and geocode), and upload the information when a communication path is present to a sound database. The earpiece is capable of running personalized sound management software disclosed herein. Moreover, hardware ingredients such as the transducers (ECR 1114, ASM 1120, and ECM 1106), processor 1116, and sealing unit 1108 are provided to manufacturers as disclosed in FIG. 3 for allowing a variety of companies to manufacture devices of their own earpiece designs. It should be noted that although an earpiece is used as an example of a personalized sound management device the components can be used in other communication devices and sound devices to personalize control of the sound.

[0122] In at least one exemplary embodiment, sealing section 1108 is a replaceable unit. Sealing section 1108 can pull out and be replaced as a unit. Sealing section 1108 performance can degrade over time due to particulate build up. There may also be sanitary or health reasons for replacing sealing section 1108 periodically. In at least one exemplary embodiment, sealing section 1108 replacement parts can be provided as part of a periodic subscription fee to a user, purchased over the counter in a store, or purchased through a web environment.

[0123] FIG. 12 is a block diagram of a device for implementing personalized sound management in accordance with at least one exemplary embodiment. A power supply 1205 powers components of the earpiece including microprocessor/DSP 1206 (or processor 1206) and a data communication system 1216. As illustrated, the earpiece can include the processor 1206 operatively coupled through a data communication system 1216 to an ASM 1210, an ECR 1212, and ECM 1208. Data communication system 1216 can include one or more Analog to Digital Converters and Digital to Analog Converters (DAC). The processor 1206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated Random Access Memory (RAM) 1202 and Read Only Memory 1204. Other memory types such as Flash, non-volatile memory, SRAM, DRAM or other like technologies can be used for storage with processor 1206. The processor 1206 can also include a clock to record a time stamp.

[0124] In general, a data communication system 1216 is a communication pathway to components of the earpiece and components external to the earpiece. The communication link can be wired or wireless. In at least one exemplary embodiment, data communication system 1216 is configured to communicate with ECM assembly 1208, ASM assembly 1210, visual display 1218, and user control interface 1214 of the earpiece. As shown, user control interface 1214 can be wired or wirelessly connected. In at least one exemplary embodiment, data communication system 1216 is capable of communication to devices exterior to the earpiece such as the user's mobile phone 1232, a second earpiece 1222, and a portable media player 1228. Portable media player 1228 can be controlled by a manual user control 1230.

[0125] The user's mobile phone includes a mobile phone 1232 communication system 1224. A microprocessor 1226 is operatively coupled to mobile phone communication system 1224. As illustrated multiple devices can be wirelessly connected to one another such as an earpiece 1220 worn by another person to the user's mobile phone 1232. Similarly, the user's mobile phone 1232 can be connected to the data communication system 1216 of the earpiece as well as the second earpiece 1222. This connection would allow one or more people to listen and respond to a call on the user's mobile phone 1232 through their respective earpieces.

[0126] As illustrated, a data communication system 1216 can include a voice operated control (VOX) module to provide voice control to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor. The VOX module can also serve as a switch to indicate to the subsystem a presence of spoken voice and a voice activity level of the spoken voice. The VOX can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the processor 1206 can provide functionality of the VOX by way of software, such as program code, assembly language, or machine

language.

[0127] ROM 1204 can be used to store personalized sound management applications to minimize the possibility of modification and tampering of the code. The RAM 1202 can also store program instructions for execution on the processor 1206 as well as captured audio processing data. For instance, memory RAM 1202 and ROM 1204 can be off-chip and external to the processor 1206 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor 1206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data. The non-volatile memory could also be used to store sonic signatures.

[0128] Data communication system 1216 can include an audio interface operatively coupled to the processor 1206 and the VOX to receive audio content, for example from portable media player 1228, cell phone 1232, or any other communication device, and deliver the audio content to the processor 1206. The processor 1206 responsive to detecting voice-operated events from the VOX can adjust the audio content delivered to the ear canal of the user of the earpiece. For instance, the processor 1206 (or the VOX of data communication system 1216) can lower a volume of the audio content responsive to detecting an event such as a sonic signature for transmitting the acute sound to the ear canal of the user. The processor 1206 by way of the ECM 1208 can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the VOX of data communication system 1216.

[0129] The earpiece and data communication system 1216 can further include a transceiver that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver can also provide support for dynamic downloading and uploading over-the-air to the earpiece. It should be noted also that next generation access technologies also can be applied to the present disclosure.

[0130] Data communication system 1216 can also include a location receiver that utilizes common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece and provide a geocode as an identifier for a recording or measurement such as sound pressure level.

[0131] The power supply 1205 can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 1205 to improve sensory input via haptic vibration. As an example, the processor 1206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0132] The earpiece can further represent a single operational device or a family of devices configured in a master-slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece can be reused in different form factors for the master and slave devices.

[0133] FIG. 13 is a diagram of a communication device 1302 or earpiece 1310 configured to provide sonic signatures to sonic signature database 1308 in accordance with at least one exemplary embodiment. Collecting a large number of sounds around the world is a daunting task. As mentioned previously, no group or business entity would have the ability to acoustically map the world on a continuous basis. In at least one exemplary embodiment, the collection of sonic signatures is achieved by mobilizing as many people as possible by making it simple to capture and provide a sound for sonic signature database 1308. This is achieved by adapting a common device having a potential of reaching billions of people for manually or automatically capturing sonic signatures and providing them to database 1308.

[0134] In at least one exemplary embodiment, communication device 1302 is a mobile communication device having a microphone for receiving sound. Examples of a communication device 1302 are a phone, cell phone, PDA, computer, two way radio, smart phone, and earpiece. Earpiece 1310 includes a microphone for capturing sound and the circuitry disclosed in FIG. 12. Earpiece 1310 is operably configured for recording sound and measuring sound pressure levels. The acoustic information received from the microphone of earpiece 1310 is recorded to memory residing in earpiece 1310 and tagged with metadata including a time stamp and geocode. The sound pressure level (SPL) is measured from the microphone signal through analog or digital signal processing typically before any audio processing occurs within earpiece 1310. The SPL measurement of this recorded event is associated to the stored recording and metadata. Similarly, communication device 1302 is adapted for capturing sound 1312 including recording, measuring sound pressure level, storing, and tagging with metadata (including a time stamp and geocode). In at least one embodiment, communication device 1302 could have additional circuitry similar to that in FIG. 12 in the device specifically for generating a sonic signature database. Alternately, the circuitry within communication device 1302 can be adapted for recording sound and measuring sound pressure level. In particular, mechanical aspects of the communication device such as microphone placement, microphone porting, as well as electronic audio processing (automatic gain control, equalization, etc. . . .) is taken into account for ensuring an accurate sound pressure level measurement and recording.

[0135] In at least one exemplary embodiment, a user having communication device 1302 manually or automatically captures and provides a sonic signature to a sonic signature database 1308. For example, a microphone on communication device 1302 is always enabled for receiving sound. The sound can be stored in a buffer. The sound in the buffer is analyzed and based on a variety of criteria can be configured to be provided to database 1308. For example, criteria such as sound pressure level, time, frequency of sound, geographic location, or recognition of a sound (sonic signature detection) are but a few of the parameters that could be used to determine that the sound in the buffer is worthy of saving. Metadata is automatically attached such as a time stamp and geocode but the user can also add information. In at least one exemplary embodiment, a communication path 1304 is opened and a link is made to website 1306 and more particularly to database 1308 where the stored sounds can be

automatically uploaded. In at least one exemplary embodiment, the sonic signature, sound pressure level, and metadata could be immediately sent if a communication path 1304 is available to save memory. Further communication between website 1308 and the user of communication device 1302 or earpiece 1310 can take place to edit, identify, describe, and format the provided sound 1312 at a more convenient time. It should be noted that video information that includes audio information can also be provided in similar fashion as disclosed hereinabove. The audio information from the video can be used for sonic signature database 1308.

[0136] Earpiece 1310 and communication device 1302 can be operably coupled together. A priority could be set up such that earpiece 1310 is the primary recorder of sound 1312 when enabled by the user. Earpiece 1310 can be used with other devices for example a portable media player. Earpiece 1310 can automatically or manually record, measure SPL, and tag metadata of sound 1312 as described hereinabove. The sonic signatures stored in earpiece 1310 could be sent to website 1306 and sonic signature database 1308 if earpiece 1310 is coupled to a communication path 1304 or through another device to which it is operably coupled that has a communication path. Alternately, the sonic signatures could be uploaded via website 1306 and sonic signature database 1308 at a more convenient time, for example a wired or wireless link to the user's personal computer at home, allowing the user to also provide additional metadata before providing the information. Thus, a common device has been provided that is adapted for capturing, storing, measuring SPL, adding metadata including a time stamp and geocode, and uploading the acoustic information to a database thereby including the broadest number of people across the largest geographic area for sound collection on a continuous basis.

[0137] FIG. 14 is a block diagram illustrating a cell phone 1404 capturing a sonic signature and providing the sonic signature to a database of sounds 1418 in accordance with at least one exemplary embodiment. Cell phone 1404 is enabled for capturing a sonic signature via an exclusive button or through an automatic process. In at least one exemplary embodiment, the user can select whether a sound is recorded, a sound pressure level measurement is recorded, or both are acquired. A default can be that both are acquired automatically.

[0138] A sound 1402 is received by a microphone on cell phone 1404 and stored in a buffer in a record sounds step 1406. The sound in the buffer is analyzed and determined to be saved. The sound pressure level (SPL) of the sonic signature is measured or calculated in a measure SPL step 1408. The user can also manually enter metadata 1410 via a keyboard to a metadata table or can enter vocal description in an attached audio stream. Metadata 1410 includes a time stamp and a geocode corresponding to the sonic signature.

[0139] In at least one exemplary embodiment, the sonic signature, sound pressure level, and metadata can be stored in memory 1412 that resides on cell phone 1404. A queue of sonic signatures 1414 can be stored in memory 1412 for uploading at an appropriate time. The user can initiate uploading of the queue of sonic signatures 1414 to database of sounds 1418 when a communication path is completed. In at least one exemplary embodiment, cell phone 1404 can automatically connect 1416 to servers in database of sounds 1418 and upload queue of sonic signatures 1414 when a communication path is enabled. Although stored on database of sounds 1418, there may be an iterative process to determine if the sonic signatures are in the correct format or are unique enough to be permanently stored. Thus, a

large database of sounds 1418 can be collected world wide by an automatic process using a common device such as a cell phone that could not be accomplished by direct means. In at least one exemplary embodiment, the database of sounds 1418 is used in conjunction with personalized sound management applications configured to provide sonic signatures for identifying and providing a response to the identified sound.

[0140] To encourage retailers to actively engage and participate in selling Hearium Labs branded products, a business model has been developed based on an annuity revenue sharing methodology. Through this patent pending approach, for the first time, retailers, operators and other suppliers will be able to carry a single audio earpiece or other device in their stores that can be sold to a wide and deep range of consumers. Utilizing an "iPod/iTunes" approach, suppliers will earn profit off the sale of the hardware in store and Personalized Sound Management applications at and after the point of sale. As the consumer continues to personalize their earpiece or device over time, the supplier will earn residuals of up to 50% of the application revenues through a co-operative sharing program between them and Hearium Labs.

[0141] To accomplish this, Hearium Labs has enabled the hardware solution to work off of a webified environment that is being architected with commercial flexibility for consumer driven point of sale purchase and activation. This approach allows for the Personalized Sound Management applications to be remotely downloaded and activated on the hardware through any Internet equipped PC and web-based browser. Consumers will thus be able to "enhance" their devices by adding both purchased and subscription applications and personalize their experience with technology designed to acclimate devices to the wearer and their environment.

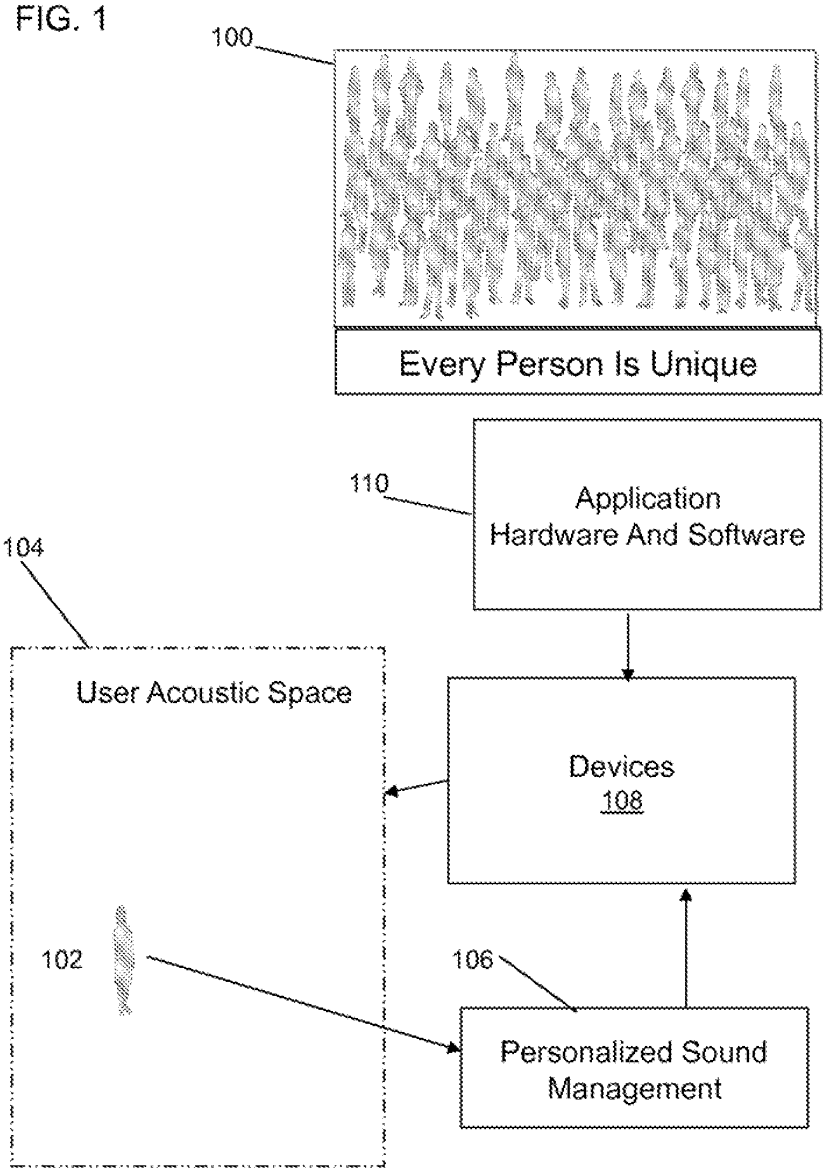
[0142] In general, a process has been provided that eliminates barriers for mass adoption of a new technology. The result is that the technology can be provided by a number of manufacturers thereby having ubiquitous availability at release. Universal compatibility is also provided by using the same hardware and software in each new product having personalized sound management. Certification ensures that each device is manufacturable and performs to specification. Manufacturers and consumers benefit because liability coverage may be provided. Consumers can adapt to the technology rapidly through an acclimation process.

[0143] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all such modifications and equivalent structures and functions.

ABSTRACT

A personalized sound management system for an acoustic space includes at least one transducer, a data communication system, one or more processors operatively coupled to the data communication system and the at least one transducer, and a medium coupled to the one or more processors. The processors access a database of sonic signatures and display a plurality of personalized sound management applications that perform at least one or more tasks among identifying a sonic signature, calculating a sound pressure level, storing metadata related to a sonic signature, monitoring sound pressure level dosage levels, switching to an ear canal microphone in a noisy environment, recording a user's voice, storing the user's voice in a memory of an earpiece device, or storing the user's voice in a memory of a server system, or converting received text received in texts or emails to voice using text to speech conversion. Other embodiments are disclosed.

FIG. 1


100
Every Person Is Unique
110
Application
Hardware And Software
104
User Acoustic Space
Devices
108
102
106
Personalized Sound
Management

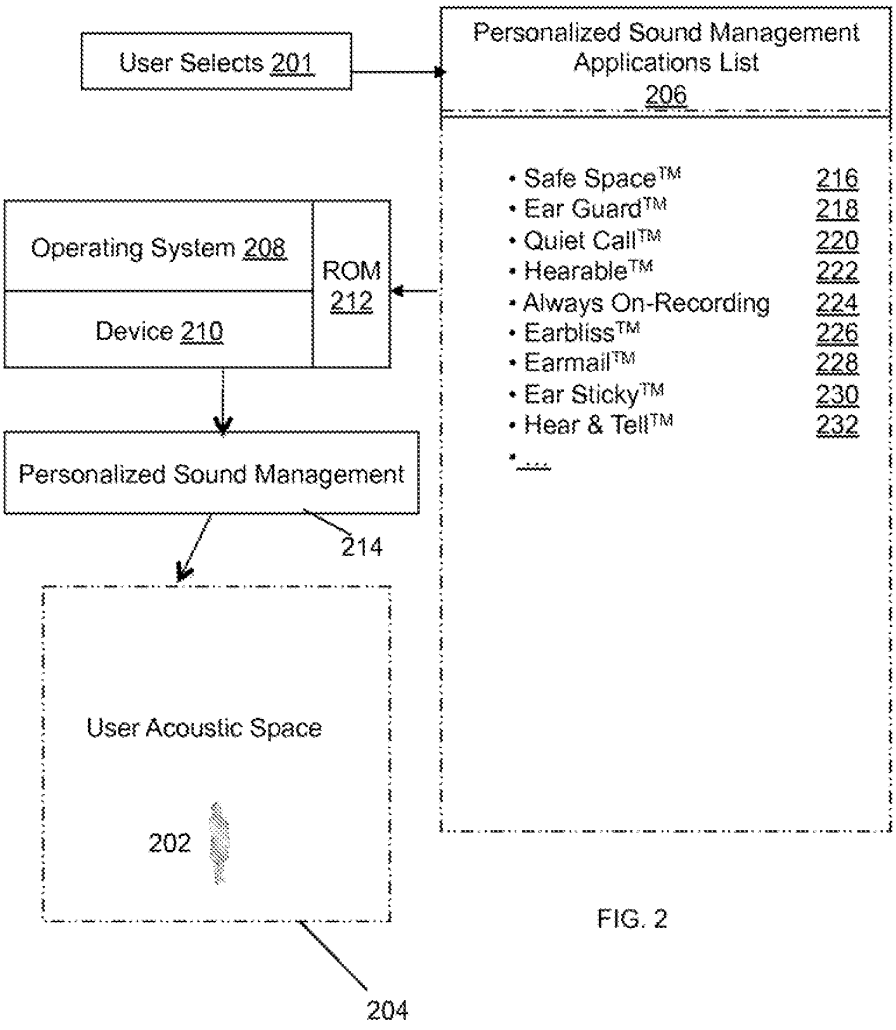

User Selects 201
Personalized Sound Management Applications List 206
• Safe Space™ 216
• Ear Guard™ 218
• Quiet Call™ 220
• Hearable™ 222
• Always On-Recording 224
• Earbliss™ 226
• Earmail™ 228
• Ear Sticky™ 230
• Hear & Tell™ 232
•....
Operating System 208
Device 210
ROM 212
Personalized Sound Management
214
User Acoustic Space
202
204

FIG. 2

FIG. 3

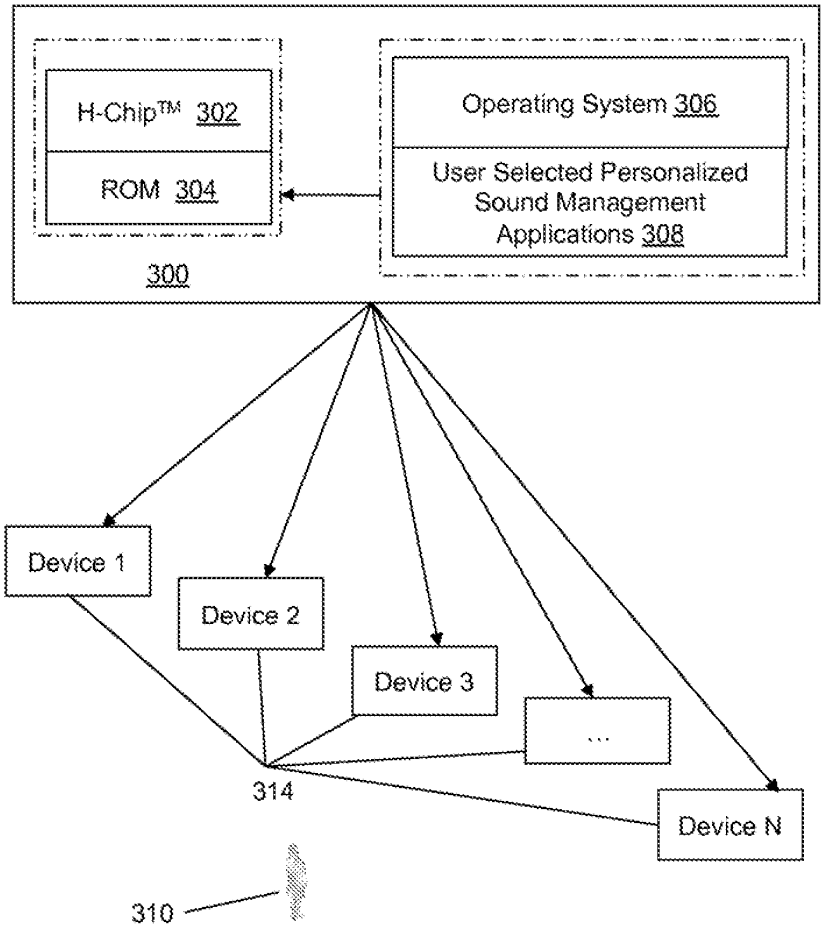

H-Chip™ 302
ROM 304
300
Operating System 306
User Selected Personalized Sound Management Applications 308
Device 1
Device 2
Device 3
...
Device N
314
310

FIG. 4

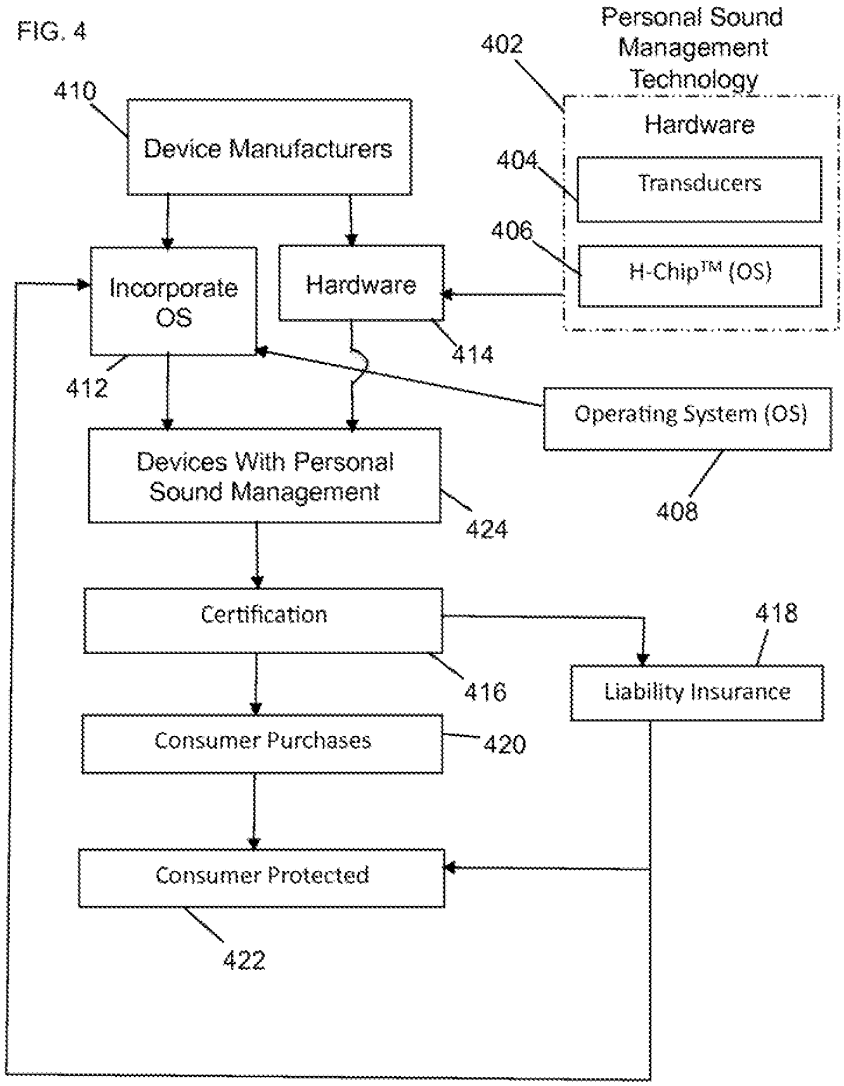
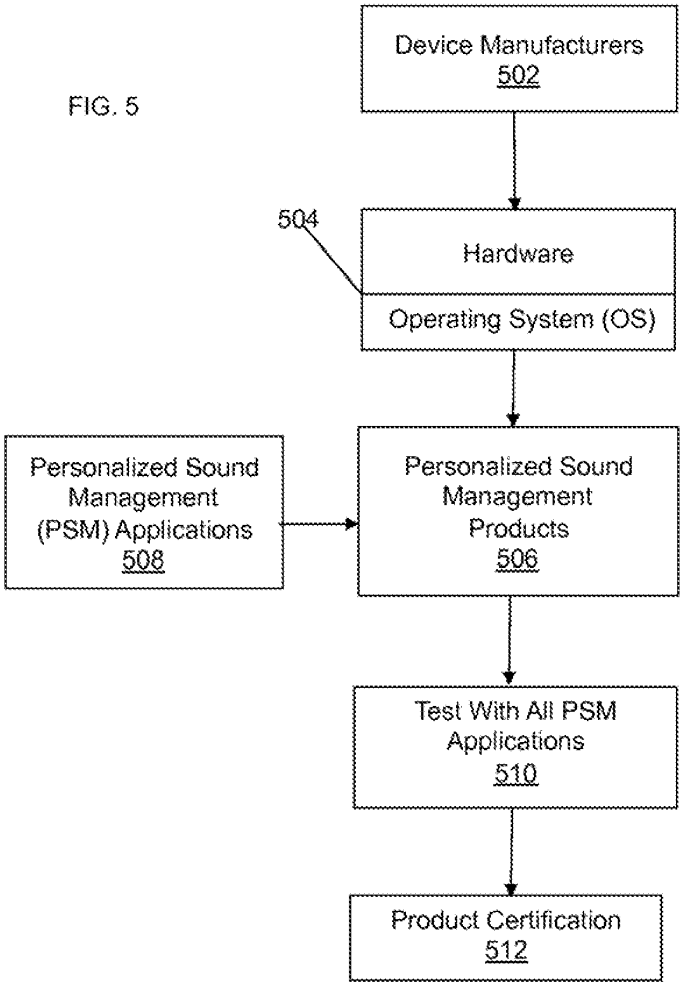

Personal Sound Management Technology
402
Hardware
404
Transducers
406
H-Chip™ (OS)
410
Device Manufacturers
Incorporate OS
Hardware
414
412
Operating System (OS)
408
Devices With Personal Sound Management
424
Certification
418
416
Liability Insurance
Consumer Purchases
420
Consumer Protected
422

FIG. 5


Device Manufacturers
502
504
Hardware
Operating System (OS)
Personalized Sound Management (PSM) Applications
508
Personalized Sound Management Products
506
Test With All PSM Applications
510
Product Certification
512

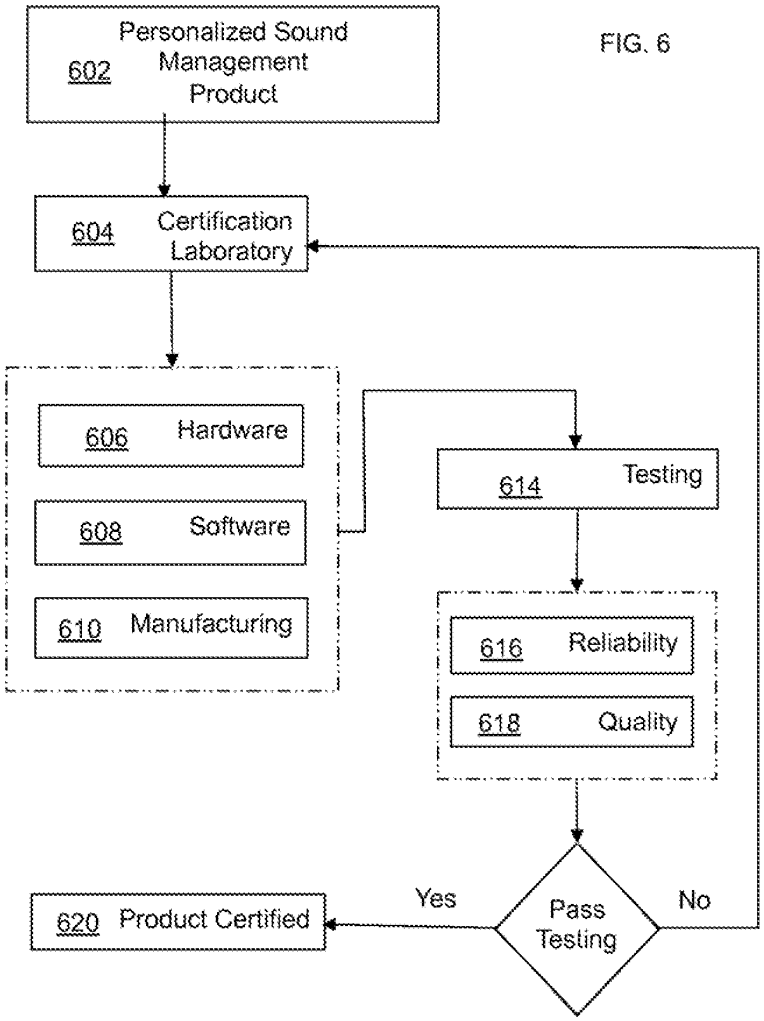
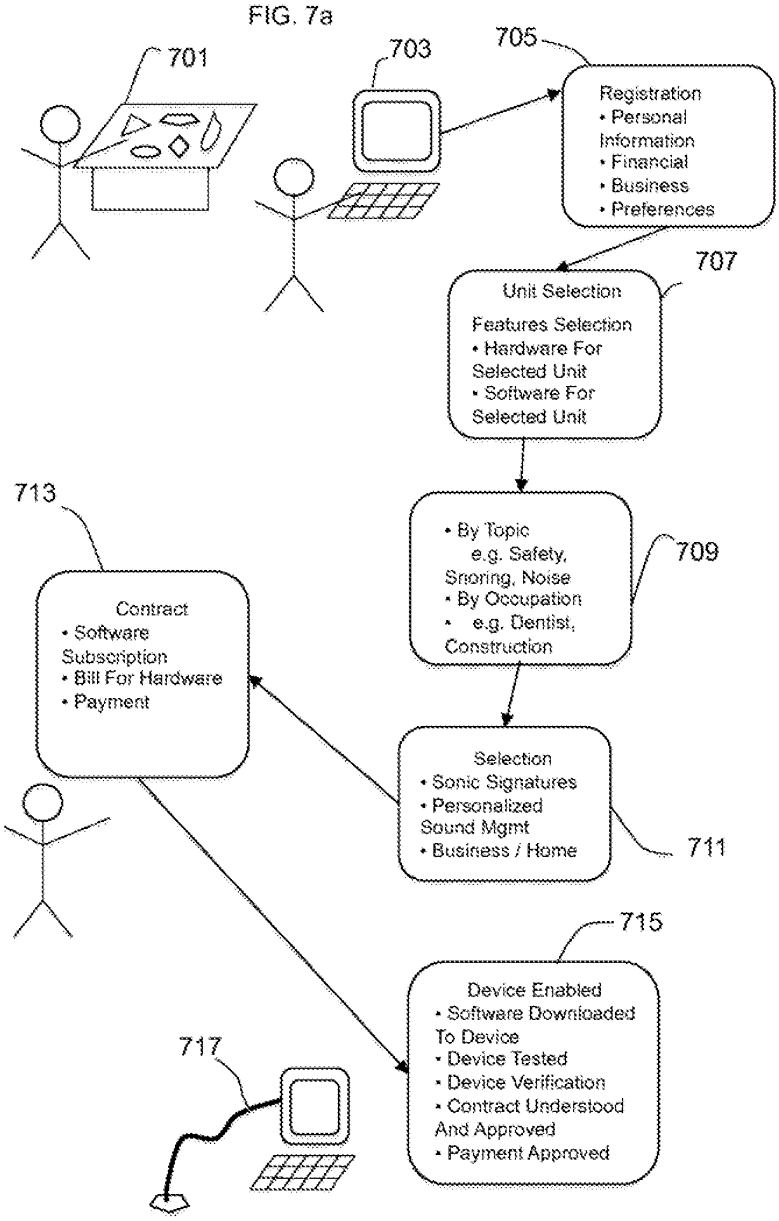

Personalized Sound Management Product
602
604 Certification Laboratory
606 Hardware
608 Software
610 Manufacturing
614 Testing
616 Reliability
618 Quality
Pass Testing
Yes
No
620 Product Certified

FIG. 6

FIG. 7a


701
703
705
Registration
• Personal Information
• Financial
• Business
• Preferences
707
Unit Selection
Features Selection
• Hardware For Selected Unit
• Software For Selected Unit
709
• By Topic
e.g. Safety, Snoring, Noise
• By Occupation
• e.g. Dentist, Construction
711
Selection
• Sonic Signatures
• Personalized Sound Mgmt
• Business / Home
713
Contract
• Software Subscription
• Bill For Hardware
• Payment
715
Device Enabled
• Software Downloaded To Device
• Device Tested
• Device Verification
• Contract Understood And Approved
• Payment Approved
717

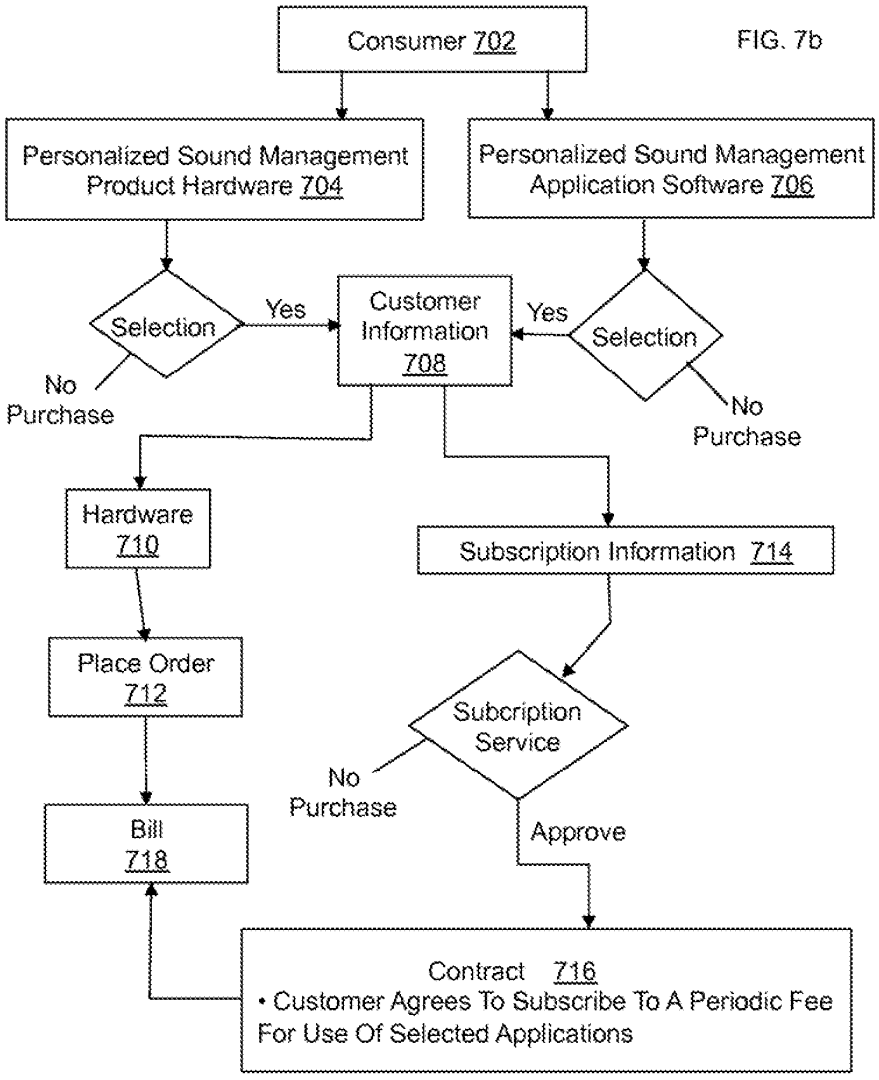

Consumer 702
Personalized Sound Management Product Hardware 704
Personalized Sound Management Application Software 706
Selection
Yes
No Purchase
Customer Information 708
Yes
Selection
No Purchase
Hardware 710
Subscription Information 714
Place Order 712
Subcription Service
No Purchase
Approve
Bill 718
Contract 716
• Customer Agrees To Subscribe To A Periodic Fee For Use Of Selected Applications

FIG. 7b

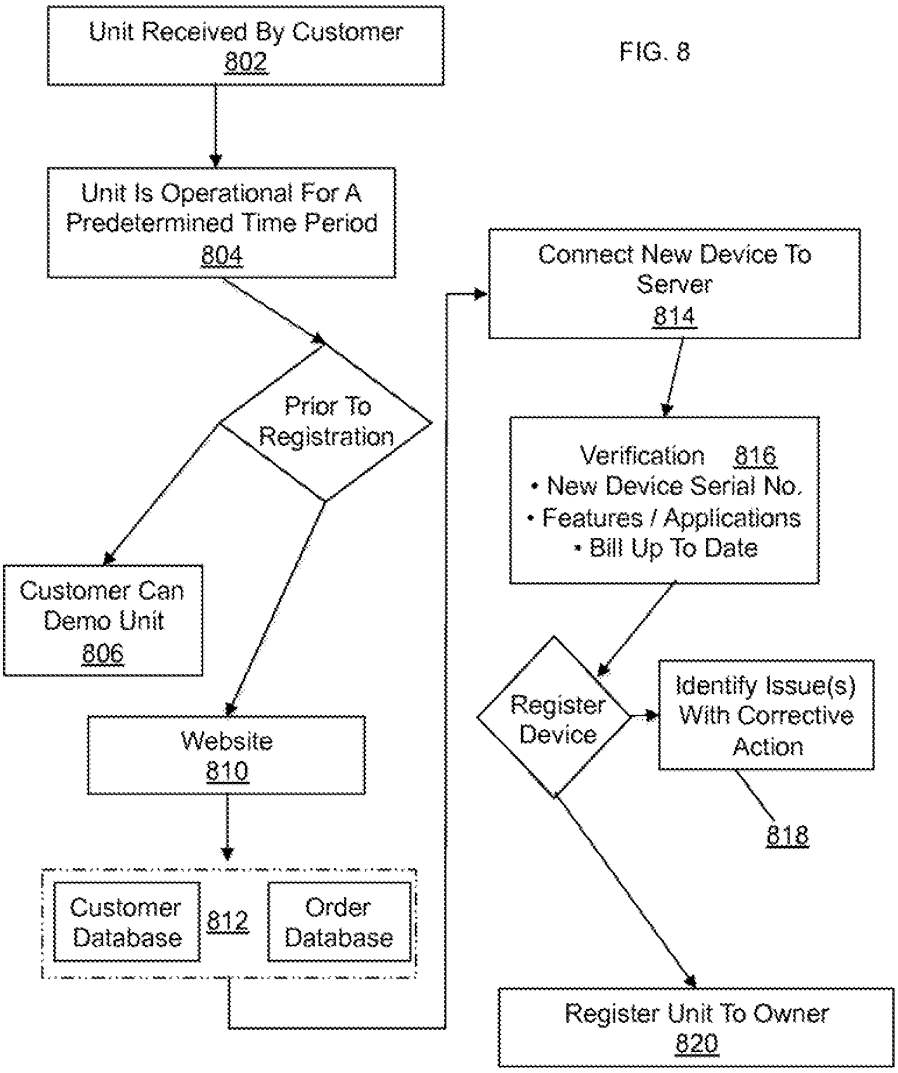
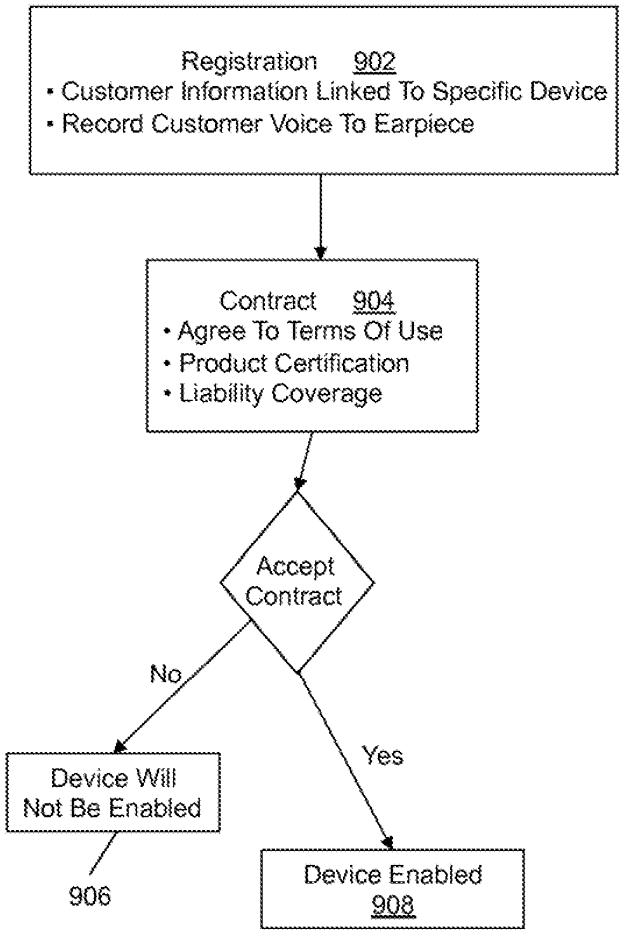

Unit Received By Customer
802
Unit Is Operational For A Predetermined Time Period
804
Prior To Registration
Customer Can Demo Unit
806
Website
810
Customer Database
812
Order Database
Connect New Device To Server
814
Verification 816
• New Device Serial No.
• Features / Applications
• Bill Up To Date
Register Device
Identify Issue(s) With Corrective Action
818
Register Unit To Owner
820

FIG. 8

FIG. 9


Registration 902
• Customer Information Linked To Specific Device
• Record Customer Voice To Earpiece
Contract 904
• Agree To Terms Of Use
• Product Certification
• Liability Coverage
Accept Contract
No
Yes
Device Will Not Be Enabled
906
Device Enabled 908

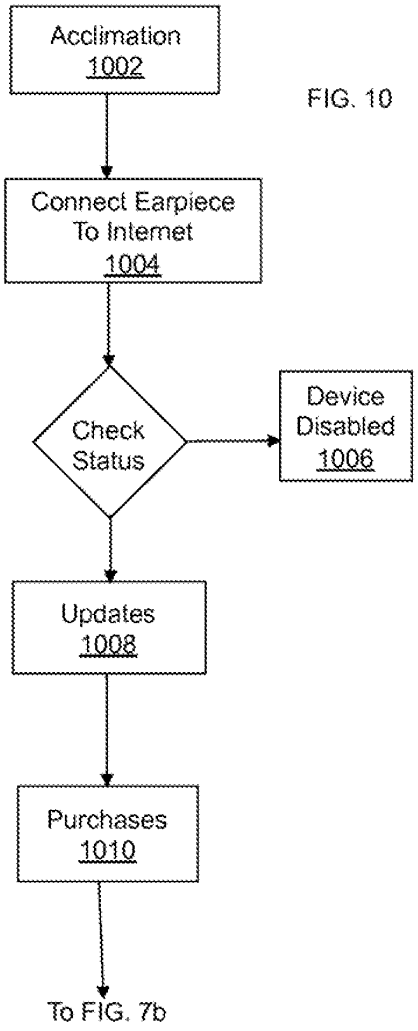

Acclimation
1002
Connect Earpiece
To Internet
1004
Check
Status
Device
Disabled
1006
Updates
1008
Purchases
1010
To FIG. 7b

FIG. 10

FIG. 11

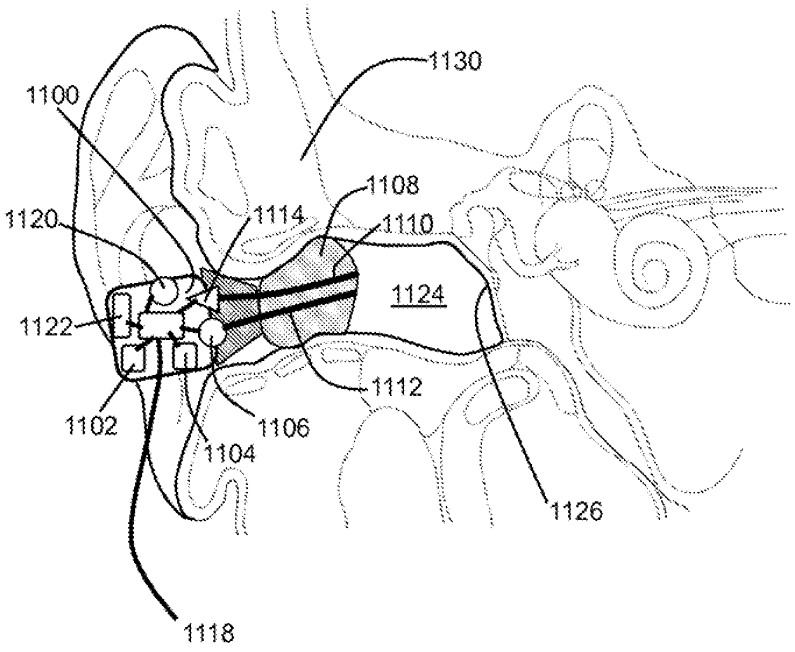

1130
1100
1108
1120
1114
1110
1124
1122
1112
1102
1106
1104
1126
1118

FIG. 12

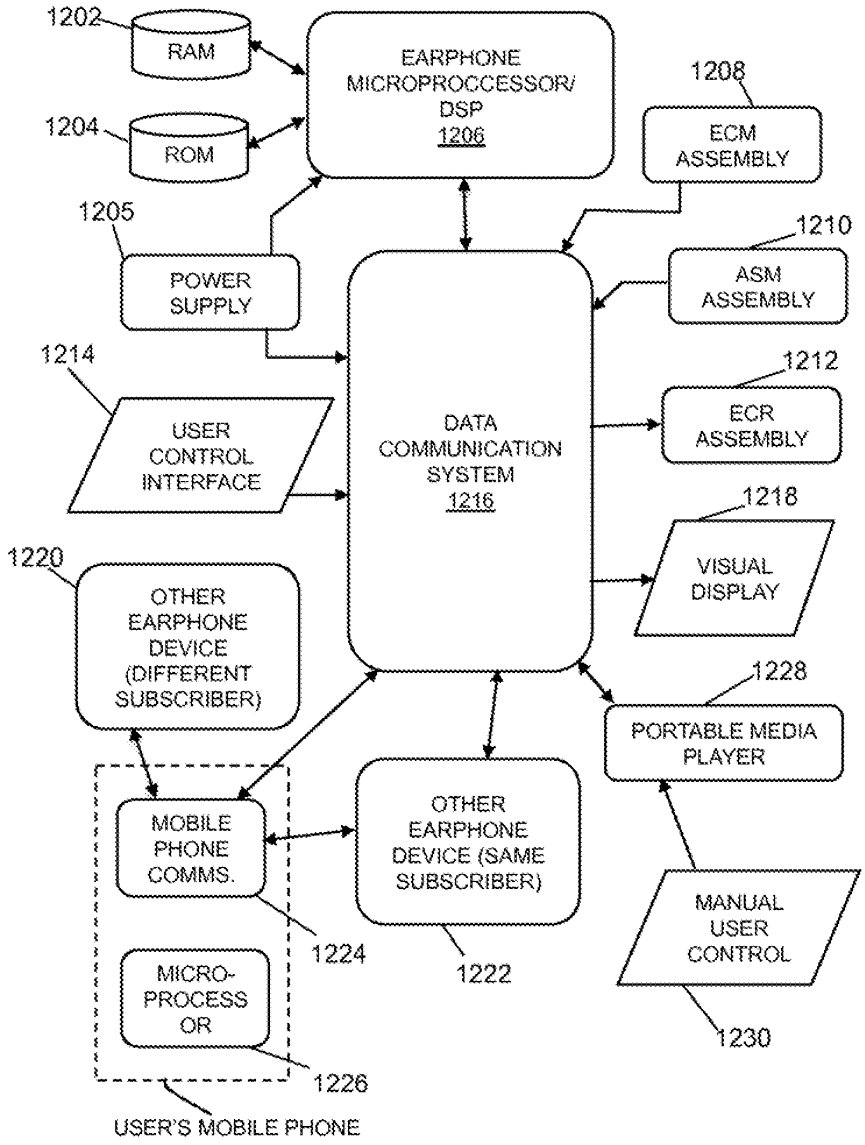

1202
RAM
1204
ROM
EARPHONE MICROPROCCESSOR/ DSP 1206
1208
ECM ASSEMBLY
1205
POWER SUPPLY
1210
ASM ASSEMBLY
1214
USER CONTROL INTERFACE
DATA COMMUNICATION SYSTEM 1216
1212
ECR ASSEMBLY
1218
VISUAL DISPLAY
1220
OTHER EARPHONE DEVICE (DIFFERENT SUBSCRIBER)
1228
PORTABLE MEDIA PLAYER
MOBILE PHONE COMMS.
OTHER EARPHONE DEVICE (SAME SUBSCRIBER)
MANUAL USER CONTROL
1224
1222
MICRO-PROCESS OR
1230
1226
USER'S MOBILE PHONE

FIG. 13

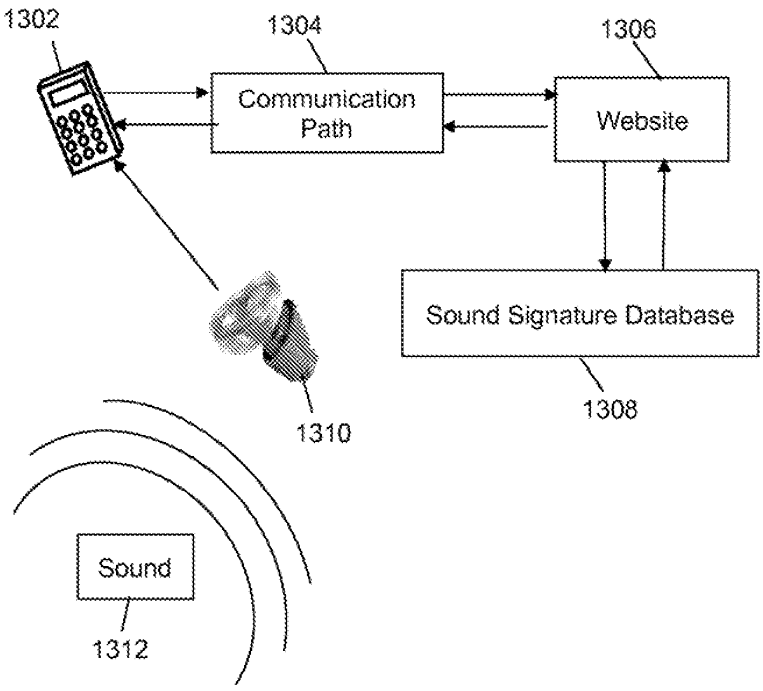
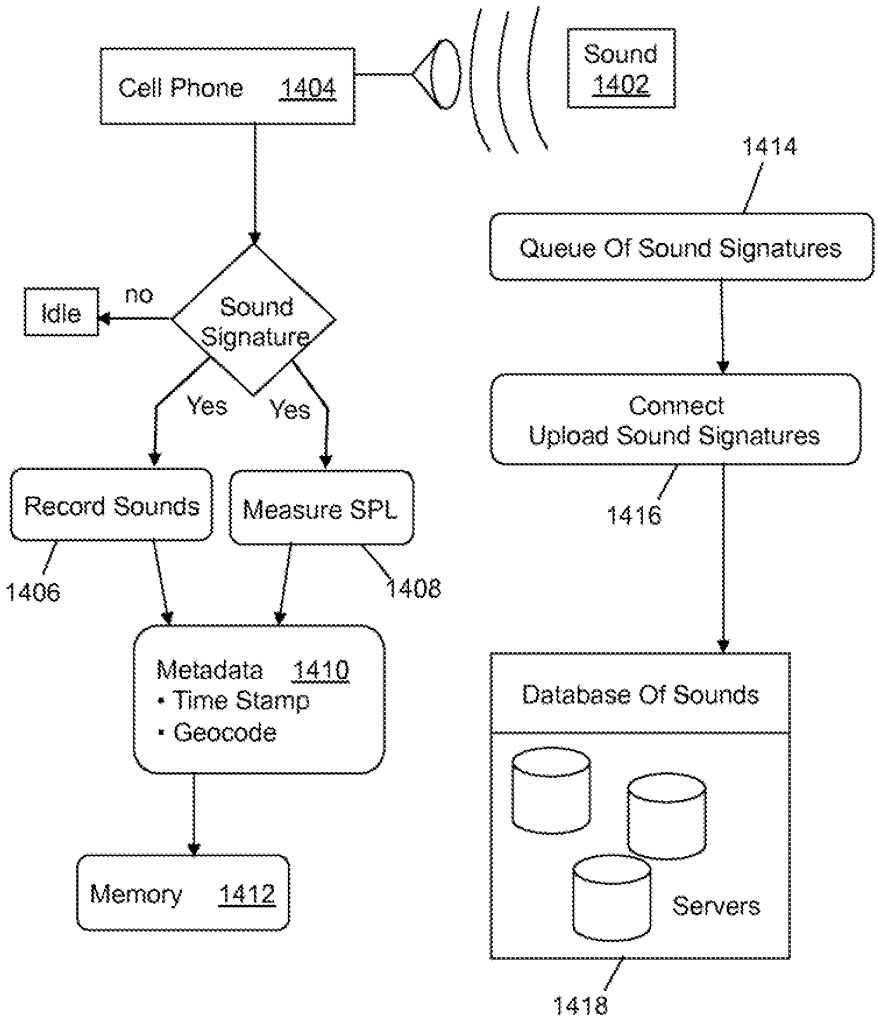

1302
1304
1306
Communication Path
Website
Sound Signature Database
1308
1310
Sound
1312

FIG. 14


Cell Phone 1404
Sound 1402
1414
Queue Of Sound Signatures
Idle
no
Sound Signature
Yes
Yes
Connect Upload Sound Signatures
Record Sounds
Measure SPL
1416
1406
1408
Metadata 1410
• Time Stamp
• Geocode
Database Of Sounds
Servers
Memory 1412
1418

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	UNKNOWN	CONFIRMATION NO. UNKNOWN
APPLICATION OF	:	STEVEN GOLDSTEIN	
DATE FILED	:		
EXAMINER	:	UNKNOWN	
TC/A.U.	:	OPAP	
DOCKET	:	OPAP	
TITLE	:	PERSONALIZED SOUND MANAGEMENT AND METHOD	

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: May 2, 2022

Sir:

The attached IDS contains the IPRs whose references have been included in a separate IDS filed herein, however in the abundance of due diligence, Applicant has also included the IPRs in the attached IDS for the Examiner's consideration. Note the Patent Owner POPR responses have not been included.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,
TECHIYA LLC

Date: May 2, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	UNKNOWN	CONFIRMATION NO. Unknown
APPLICATION OF	:	Steven Goldstein	
DATE FILED	:		
EXAMINER	:	UNKNOWN	
TC/A.U.	:	SUGGESTED 2654	
DOCKET	:	CONT-17203731	
TITLE	:	PERSONALIZED SOUND MANAGEMENT AND METHOD	

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: May 3, 2022

Sir:

Applicant, submits the attached continuation application of U.S. Patent Application No. 17203731 to which the attached continuation application claims priority to. Additionally, Applicant has attempted to include all references that were considered in previous prosecutions of the subject matter and any relevant IPRs, however Applicant appreciates if the Examiner would also look at any reference that may have been missed in the attached IDSs that were examined in the family of U.S. Patent Application No. 17203731.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

Date: May 3, 2022

/John P. Keady/
John P. Keady
Ph.D (Physics), Ph.D (Comp Sci.), MBA, JD,
VA Bar 47027, Pat. Reg. No. 56389
CTO and Assistant General Counsel of IP
571-606-1291

AMENDMENTS TO THE CLAIMS

The following listing of claims will replace all prior versions, and listings, of claims in the application.

Listing of Claims:

1.(currently amended) An earpiece comprising:

a sealing unit;

a first microphone generating a first signal, wherein the first microphone is configured to measure a first acoustic environment;

a second microphone generating a second signal, wherein the second microphone is configured to measure a second acoustic environment, wherein the second acoustic environment is closer to a user's ear canal than the first acoustic environment;

a speaker;

a memory that stores instructions;

a circular buffer, where the circular buffer is configured to store a portion of at least one of the first signal, the second signal or a combination thereof, generating data stored in the circular buffer; and

a microprocessor, wherein the microprocessor is operatively connected to the first microphone, the second microphone, the speaker, the circular buffer, and the memory,

wherein the microprocessor is configured to execute the instructions to perform operations comprising:

sending an audio content signal to the speaker;

analyzing the data to detect an event; and

reducing a volume of the audio content signal, for a predetermined time period, when the event is detected, wherein during the predetermined time period the microprocessor is still analyzing the data to detect [[the]] a second event.

2.(original) The earpiece according to claim 1, wherein the event is a user's voice.

3.(original) The earpiece according to claim 1, wherein the event is a voice.

4.(previously presented) The earpiece according to claim 2, wherein the sound level of the audio content is reduced for a predetermined time period after the detection of a user's voice.

5.(previously presented) The earpiece according to claim 4, where the predetermined time period is selected by the user.

6.(original) The earpiece according to claim 1, where the event is an emergency event.

7.(original) The earpiece according to claim 6, where the emergency event is at least one of a siren, an emergency sound, or an alarm or a combination of these.

8.(previously presented) The earpiece according to 1, wherein detecting an event compares the data to a sonic signature stored in the memory.

9.(previously presented) The earpiece according to claim 8, wherein the event is detected when the comparison indicates that the data is within a threshold value of the sonic signature.

10.(original) The earpiece according to claim 1, where the first microphone is an ambient sound microphone.

11.(original) The earpiece according to claim 10, where the second microphone is a second ambient sound microphone.

12.(original) The earpiece according to claim 10, where the second microphone is an ear canal microphone.

13.(previously presented) The earpiece according to claim 5, where the predetermined time period is the time after which no further user voice is detected.

14.(original) The earpiece according to claim 13, wherein the first microphone is an ambient sound microphone.

15.(original) The earpiece according to claim 14, where the second microphone is a second ambient sound microphone.

16.(original) The earpiece according to claim 14, where the second microphone is an ear canal microphone.

17.(previously presented) The earpiece according to claim 13, where the user selects the predetermined time period using a user control interface on a communication device.

REMARKS

Applicant appreciates the indication of allowance of claims 1-17, however respectfully submits a narrowing amendment for claim 1. For at least the reasons given above, amended claim 1, and claim 2-17, which depend thereupon are allowable.

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	
Filing Date	
First Named Inventor	Goldstein
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-17203731

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
	5	7562020		2009-06-14	Le et al.	
	6	9135797		2015-09-15	Couper et al.	
	7	9123343		2015-09-01	Kurki-Suonio	
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	First Named Inventor	Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

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	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
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	27	7920557		2011-04-05	Moote	
	28	7158933		2007-01-02	Balan; Radu Victor	
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	1	20030165246		2003-09-04	Kvaloy et al.	
	2	20050288057		2005-12-29	Lai et al.	
	3	20060195322		2006-08-31	Broussard et al.	
	4	20070043563		2007-02-22	Comerford et al.	
	5	20010046304		2001-11-29	Rast	
	6	20070189544		2007-08-16	Rosenberg	
	7	20060092043		2006-05-04	Lagassey	
	8	20020106091		2002-08-08	Furst et al.	
	9	20040109668		2004-06-10	Stuckman	
	10	20020118798		2002-08-29	Langhart et al.	

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	11	20040042103		2004-03-04	Mayer	
	12	20030161097		2003-08-28	Le et al.	
	13	20160104452		2016-04-14	Guan et al.	
	14	20040203351		2004-10-14	Shearer et al.	
	15	20080165988		2008-07-10	Terlizzi et al.	
	16	20070291953		2007-12-20	Ngia et al.	
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	20	20090010456		2009-01-08	Goldstein et al.	
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	22	20040190737		2004-09-30	Kuhnel et al.	
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	27	20110264447		2011-10-27	Visser et al.	
	28	20100296668		2010-11-25	Lee et al.	
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	33	20090286515		2009-05-05	Othmer; Konstantin		
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	1	EP1519625	EP	A2		Victorian		
	2	2006037156	WO	A1		Mejia et al.		

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	1	Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan 9-12, 2005, p. 284-286.	
	2	Bernard Widrow, John R. Glover Jr., John M. McCool, John Kaunitz, Charles S. Williams, Robert H. Hearn, James R. Zeidler, Eugene Dong Jr, and Robert C. Goodlin, Adaptive Noise Cancelling: Principles and Applications, PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975	

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	3	Mauro Dentino, John M. McCool, and Bernard Widrow, Adaptive Filtering in the Frequency Domain, PROCEEDINGS OF THE IEEE, VOL. 66, NO. 12, DECEMBER 1978	

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	1	5852804		1998-12-22	Sako	
	2	6618073		2003-09-09	Lambert et al.	
	3	6754359		2004-06-22	Svean et al.	
	4	7072476		2006-07-04	White et al.	
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	9	8600454		2013-12-03	Nicholson	
	10	8583428		2013-11-12	Tashev et al.	
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	15	8391501		2013-03-05	Khawand et al.	
	16	4237343		1980-12-02	Kurtin et al.	
	17	8625819		2014-01-07	Goldstein et al.	
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	20	7346176		2008-03-18	Bernardi et al.	
	21	7672845		2010-03-02	Beranek et al.	
	22	9961435		2018-05-01	Goyal et al.	
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	27	10884696		2021-01-05	Willis; Benjamin S.	

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	1	20090010444		2009-01-08	Goldstein et al.	

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	2	20070021958		2007-01-25	Visseret al.	
	3	20070033029		2007-02-08	Sakawaki	
	4	20070086600		2007-04-19	Boesen	
	5	20070098192		2007-05-03	Sipkema	
	6	20070237341		2007-10-11	Laroche	
	7	20070291953		2007-12-20	Ngia et al.	
	8	20080037801		2008-02-14	Alves et al.	
	9	20080187163		2008-08-07	Goldstein et al.	
	10	20080317259		2008-12-25	Zhang et al.	
	11	20110135107		2011-06-09	Konchitsky	
	12	20070230712		2007-10-04	Belt et al.	

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	13	20070088544		2007-04-19	Acero et al.	
	14	20070076898		2007-04-05	Sarroukh et al.	
	15	20080147397		2008-06-19	Konig et al.	
	16	20070076896		2007-04-05	Hosaka et al.	
	17	20070053522		2007-03-08	Murray et al.	
	18	20060074693		2006-04-06	Yamashita	
	19	20060133621		2006-06-22	Chen et al.	
	20	20050175189		2005-08-11	Lee et al.	
	21	20030035551		2003-02-20	Light et al.	
	22	20090209290		2009-08-20	Chen et al.	
	23	20050058313		2005-03-17	Victorian et al.	

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	24	20080137873		2008-06-12	Goldstein	
	25	20080181419		2008-07-31	Goldstein et al.	
	26	20110206217		2011-08-25	Wels	
	27	20080260180		2008-10-23	Goldstein et al.	
	28	20070189544		2007-08-16	Rosenberg	
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	30	20170142511		2017-05-18	Dennis	
	31	20070049361		2007-03-01	Coote et al.	
	32	20180206025		2018-07-19	Rule Allen John	
	33	20190069069		2019-02-28	Radin; Daniel I.	
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	1	2012078670	WO		2012-06-14	Yousefian et al.		
	2	2007085307	WO		2007-08-02	Sony Ericsson Mobile Communications Ab		
	3	5639160	JP		2014-12-10			

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Name/Print	John P. Keady	Registration Number	56389

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7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

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9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

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POWER OF ATTORNEY TO PROSECUTE APPLICATIONS BEFORE THE USPTO

I hereby revoke all previous powers of attorney given in the application identified in the attached statement under 37 CFR 3.73(b).

I hereby appoint:

☑ Practitioners associated with the Customer Number: 150139

OR

☐ Practitioner(s) named below (if more than ten patent practitioners are to be named, then a customer number must be used):

Name	Registration Number	Name	Registration Number

as attorney(s) or agent(s) to represent the undersigned before the United States Patent and Trademark Office (USPTO) in connection with any and all patent applications assigned only to the undersigned according to the USPTO assignment records or assignment documents attached to this form in accordance with 37 CFR 3.73(b).

Please change the correspondence address for the application identified in the attached statement under 37 CFR 3.73(b) to:

☑ The address associated with Customer Number: 150139

OR

☐ Firm or Individual Name			
Address			
City		State	Zip
Country			
Telephone		Email	

Assignee Name and Address:

Staton Techiya LLC. 16193 Bridlewood Circle, Delray Beach, FL 33445

A copy of this form, together with a statement under 37 CFR 3.73(b) (Form PTO/SB/96 or equivalent) is required to be filed in each application in which this form is used. The statement under 37 CFR 3.73(b) may be completed by one of the practitioners appointed in this form if the appointed practitioner is authorized to act on behalf of the assignee, and must identify the application in which this Power of Attorney is to be filed.

SIGNATURE of Assignee of Record

The individual whose signature and title is supplied below is authorized to act on behalf of the assignee

Signature		Date	8-14-17
Name	DANIEL C - STATON	Telephone	513-404-0099
Title	MANAGING MEMBER		

This collection of information is required by 37 CFR 1.31, 1.32 and 1.33. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 3 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

Title of Invention	Personalized Sound Management and Method

As the below named inventor, I hereby declare that:

This declaration is directed to:

☐ The attached application, or

☒ United States application or PCT international application number 14/846,994

filed on 09-07-2015.

The above-identified application was made or authorized to be made by me.

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

WARNING:

Petitioner/applicant is cautioned to avoid submitting personal information in documents filed in a patent application that may contribute to identity theft. Personal information such as social security numbers, bank account numbers, or credit card numbers (other than a check or credit card authorization form PTO-2038 submitted for payment purposes) is never required by the USPTO to support a petition or an application. If this type of personal information is included in documents submitted to the USPTO, petitioners/applicants should consider redacting such personal information from the documents before submitting them to the USPTO. Petitioner/applicant is advised that the record of a patent application is available to the public after publication of the application (unless a non-publication request in compliance with 37 CFR 1.213(a) is made in the application) or issuance of a patent. Furthermore, the record from an abandoned application may also be available to the public if the application is referenced in a published application or an issued patent (see 37 CFR 1.14). Checks and credit card authorization forms PTO-2038 submitted for payment purposes are not retained in the application file and therefore are not publicly available.

LEGAL NAME OF INVENTOR

Inventor: John P. Keady Date (Optional) :

Signature: /John P. Keady/

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

This collection of information is required by 35 U.S.C. 115 and 37 CFR 1.63. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 1 minute to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

Privacy Act Statement

The **Privacy Act of 1974 (P.L. 93-579)** requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether disclosure of these records is required by the Freedom of Information Act.
2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.
3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.
4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).
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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

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This declaration is directed to:

☐ The attached application, or

☒ United States application or PCT international application number 14/846,994

filed on 09-07-2015.

The above-identified application was made or authorized to be made by me.

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LEGAL NAME OF INVENTOR

Inventor: Steven Goldstein Date (Optional) :

Signature: /Steven Goldstein/

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

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Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

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	Application Number	
	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	
	Examiner Name	UNKNOWN
	Attorney Docket Number	CIP-17203731

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
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	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	
	Examiner Name	UNKNOWN
	Attorney Docket Number	CIP-17203731

	1	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00282, December 21, 2021	
	2	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00242, December 23, 2021	
	3	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00243, December 23, 2021	
	4	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00234, December 21, 2021	
	5	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00253, January 18, 2022	
	6	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00324, January 13, 2022	
	7	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00281, January 18, 2022	
	8	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00302, January 13, 2022	
	9	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00369, February 18, 2022	
	10	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00388, February 18, 2022	
	11	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00410, February 18, 2022	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	
	Examiner Name	UNKNOWN
	Attorney Docket Number	CIP-17203731

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Examiner Signature		Date Considered	

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through a citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	
	Examiner Name	UNKNOWN
	Attorney Docket Number	CIP-17203731

CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-04-29
Name/Print	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

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PTO/SB/08a (02-18)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	
Filing Date	
First Named Inventor	Goldstein
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT-17203731

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
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	First Named Inventor	Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	

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	1	20030165246		2003-09-04	Kvaloy et al.	

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	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

	2	20080037801		2008-02-14	Alves et al.		
	3	20060204014		2006-09-14	Isenberg et al.		
	4	20050078838		2005-04-14	Simon		

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EXAMINER SIGNATURE

Examiner Signature		Date Considered	

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through a citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
	Filing Date	
	First Named Inventor	Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT-17203731

[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

☒ A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-05-02
Name/Print	John P. Keady	Registration Number	56389

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

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7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

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