(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
8 August 2013 (08.08.2013)

WIPO | PCT

(10) International Publication Number
WO 2013/115979 A1

(51) International Patent Classification:
H04M 1/60 (2006.01) *H03G 3/32* (2006.01)

(21) International Application Number:
PCT/US2013/021552

(22) International Filing Date:
15 January 2013 (15.01.2013)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
13/365,390 3 February 2012 (03.02.2012) US

(71) Applicant: **MOTOROLA MOBILITY LLC** [US/US]; 600 North US Highway 45, Libertyville, Illinois 60048 (US).

(72) Inventors: **ZUREK, Robert, A.**; 1055 Autumn Drive, Antioch, Illinois 60002 (US). **ALAMEH, Rachid, M.**; 4919 Daniel Drive, Crystal Lake, Illinois 60014 (US). **DICKINSON, Timothy**; 1572 Rolling Hills, Crystal Lake, Illinois 60014 (US). **MERRELL, Thomas, Y.**; 39855 North Ackworth Lane, Beach Park, Illinois 60083 (US).

(74) Agents: **CHEN, Sylvia** et al.; 600 North US Highway 45, Libertyville, Illinois 60048 (US).

(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BN, BR, BW, BY, BZ, CA, CH, CL, 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, PA, PE, PG, PH, PL, PT, QA, RO, RS, RU, RW, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TH, 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, LR, LS, MW, MZ, NA, RW, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, RU, TJ, TM), European (AL, AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MK, MT, NL, NO, PL, PT, RO, RS, SE, SI, SK, SM, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

[Continued on next page]

(54) Title: MOTION BASED COMPENSATION OF UPLINKED AUDIO

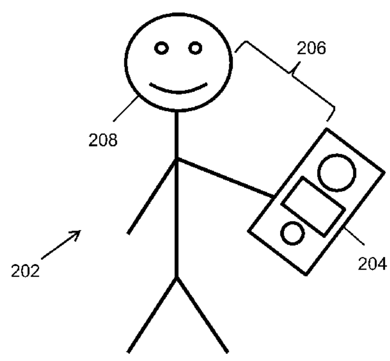

Fig. 2

(57) Abstract: Embodiments relate to apparatuses for, and methods of, compensating for distance 206 and velocity present between a microphone and user's 202 mouth. Such devices and methods allow compensation for varying amplitudes of sound pressure received due to a varying distance 206 between a microphone and an originating user's 202 mouth. Additionally, the devices and methods may also compensate for pitch shifts due to a Doppler effect caused by a non-zero velocity of a microphone relative to a user's 202 mouth.

WO 2013/115979 A1

Published:

— *with international search report (Art. 21(3))*

MOTION BASED COMPENSATION OF UPLINKED AUDIO

CROSS-REFERENCE TO RELATED APPLICATION

[0001] The present application is related to co-pending U.S. utility patent application no. 13/365,387 entitled "MOTION BASED COMPENSATION OF DOWNLINKED AUDIO," by Robert A. Zurek et al. (CS38847) filed on February 3, 2012, and the contents thereof are hereby incorporated by reference herein in its entirety.

FIELD

[0002] The present teachings relate to systems for, and methods of, compensating for a varying distance between a microphone in a mobile electronic device and a user's mouth.

DESCRIPTION OF DRAWINGS

[0003] The accompanying drawings, which are incorporated in and constitute a part of this specification, illustrate embodiments of the present teachings and together with the description, serve to explain the principles of the present teachings. In the figures:

[0004] FIG. 1 is a schematic diagram of a mobile device according to various embodiments;

[0005] FIG. 2 is a schematic diagram of user interacting with a mobile device according to various embodiments;

[0006] FIG. 3 is a flow chart depicting a method of motion based compensation of downlinked audio according to various embodiments;

[0007] FIG. 4 is a flow chart depicting a method of motion based compensation of uplinked audio according to various embodiments;

[0008] FIG. 5 is a flow chart depicting a method of intuitive motion based microphone gain adjustment according to various embodiments;

[0009] FIG. 6 is a flowchart depicting a method of noise abatement in uplinked audio according to various embodiments; and

[0010] FIG. 7 is a flowchart depicting a method of compensating for a Doppler effect in uplinked audio according to various embodiments.

DESCRIPTION OF EMBODIMENTS

[0011] Techniques compensate for the effect of a varied distance, and relative movement, between a microphone in a mobile device and the mouth of a user. In general, as a distance between a microphone and a user's mouth increases, the sound pressure of detected audio decreases (correspondingly, as distance decreases, detected sound pressure increases). The relative distance may change due to movement of the user's head, the device, or both. Certain embodiments compensate for this effect by adjusting a gain of a microphone amplifier in proportion to the distance. Furthermore, certain embodiments compensate for increased noise due to increased amplifier gain by dynamically adjusting a noise reducing filter. Certain embodiments also compensate for a Doppler effect produced by a relative velocity between the microphone of a device and the user's mouth. Certain embodiments also allow a user to intuitively and efficiently adjust a gain of the microphone in the mobile device by activating a microphone gain set mode. When in the microphone gain set mode, the user may move the mobile device toward or away from his or her head and the gain level will be adjusted in inverse proportion to the distance. The device may be mobile, such as a cellular telephone according to certain embodiments. In some embodiments, the device may be a speakerphone.

[0012] According to various embodiments, a method compensates for movement of a microphone relative to a user's head, where the microphone is present in a mobile device. The method includes producing, by the device, an electrical signal representative of audio received at the microphone and determining, by the device, a distance between the device and the user's head. The method also includes automatically setting, by the device, a gain of the electrical signal in accordance with the distance. The method may further include modifying, by the device, an audio filtering in accordance with the distance, wherein the audio filtering is applied to the electrical signal. The method may further include generating, by the device, an output signal representative of the audio with the gain and the audio filtering.

[0013] Reference will now be made in detail to exemplary embodiments of the present teachings, which are illustrated in the accompanying drawings. Where possible the same reference numbers will be used throughout the drawings to refer to the same or like parts.

[0014] FIG. 1 is a schematic diagram of a device according to various embodiments. Lines between blocks in FIG. 1 indicate communicative coupling and do not necessarily represent direct continuous electrical connection. The device 102 may be, by way of non-limiting example, a mobile device, a cellular telephone, a recorded audio player (e.g., a MP3 player), a personal digital assistant, a tablet computer, or other type of hand-held or wearable computer, telephone, or device containing a loudspeaker or microphone. Mobile device 102 includes processor 104. Processor 104 may be, by way of non-limiting example, a microprocessor or a microcontroller. Processor 104 may be capable of carrying out electronically stored program instructions. Processor 104 may contain or be coupled to timer 124. Processor 104 may be coupled to antenna 126. Processor 104 may be communicatively coupled to persistent memory 110. Persistent memory 110 may include, by way of non-limiting example, one or both of a hard drive and a flash memory device. Persistent memory 110 may store instructions which, when executed by processor 104 in conjunction with other disclosed elements, constitute systems and perform methods disclosed herein.

[0015] Processor 104 may be further coupled to display 106 and other user interface 108 elements. Display 106 may be, by way of non-limiting example, a liquid crystal display, which may include a touchscreen. Other user interface 108 elements may be, by way of non-limiting example, a full or partial physical keyboard or keypad. In embodiments where display 106 is a touchscreen, display 106 may be combined with user interface 108 so as to display an active full or partial keyboard or keypad. That is, user interface 108 may include a full or partial virtual keyboard or keypad.

[0016] Processor 104 may be further coupled to loudspeaker 114 by way of amplifier 112. Loudspeaker 114 may be, by way of non-limiting example, a loudspeaker of a cellular telephone or audio system. Loudspeaker 114 may be capable of producing sound suitable for a speakerphone mode or a private telephone mode. Amplifier 112 may include a preamplification stage and a power amplification stage. In some embodiments, amplifier 112 may include one or both of a digital-to-analog converter and decoding (e.g., compression, decompression, and/or error correction decoding) circuitry.

[0017] Processor 104 may be further coupled to microphone 118 by way of amplifier 116. Microphone 118 may be, by way of non-limiting example, a microphone of a cellular telephone. Microphone 118 may be capable of receiving and converting to electricity sound captured by the cellular telephone. Amplifier 116 may include a pre-amplification stage. In some embodiments, amplifier 116 may include one or both of an analog-to-digital converter and encoding (e.g., error correction and/or compression encoding) circuitry.

[0018] Processor 104 may be further coupled to sensor system 120. Sensor system 120 may be any of several various types. By way of non-limiting example, sensor system 120 may be infrared, acoustic, or photographic. If infrared, sensor system 120 may include an infrared emitter (e.g., a high-power light emitting diode) and an infrared receiver (e.g., an infrared sensitive diode). If acoustic, sensor system 120 may include an ultrasonic transducer or separate ultrasonic emitters and receivers. In some embodiments, microphone 118 may perform ultrasonic reception. If photographic, sensor system 120 may include a camera utilizing, e.g., optics and a charge coupled device. In some embodiments in which sensor system 120 is photographic, one or both of sensor system 120 and processor 104 may employ facial recognition, known to those of skill in the art, capable of determining when a human face is within a depth of field of sensor system 120. Regardless as to the particular technology used by sensor system 120, sensor system 120 may include interpretive circuitry that is capable of converting raw empirical measurements into electrical signals interpretable by processor 104.

[0019] Sensor system 120 may further include accelerometer 122, which detects applied linear force (e.g., in one, two or three linearly orthogonal directions). Accelerometer 122 may be, by way of non-limiting example, a micro-electromechanical system (MEMS), capable of determining the magnitude and direction of any acceleration. Sensor system 120 may also include a gyroscope (possibly as, or as part of, accelerometer 122) that detects applied rotational force (e.g., in one, two or three rotationally orthogonal directions). Sensor system 120 may further include a velocity sensor, which detects the velocity of objects relative to a face of the mobile device 102. The velocity sensor may be, by way of non-limiting example, an optical interferometer capable of determining the magnitude and direction of any velocity of the device relative to an object in front of the sensor. The velocity sensor may detect velocity only in a direction normal (i.e., perpendicular) to the face (e.g., display) of the mobile device, or in three orthogonal directions.

[0020] FIG. 2 is a schematic diagram of a user interacting with a mobile device according to various embodiments. In particular, user 202 is depicted as holding mobile device 204, which may be, by way of non-limiting example, mobile device 102 of FIG. 1. User 202 may interact with mobile device by one or both of providing audio input (e.g., voice) and receiving audio output (e.g., audio provided by the device 102). Note that there may not be a consistent distance 206 between the mobile device 204 and the user 202. For a handheld mobile device 204 as depicted, the distance may vary from moment to moment depending on the angle of the hand, wrist, elbow, shoulder, neck, and head of the user. Also, the user may shift the mobile device 204 from one hand to another, put the mobile device 204 down on a table and pace while talking and listening, and many other physical interactions that affect the distance between the mobile device 204 and the user 202 which in turn affect the sound pressure from the loudspeaker of the device as detected by the user's ear(s) as well as the sound pressure produced from the user's mouth as detected by the microphone of the device.

[0021] Mobile device 204 is capable of detecting a distance 206 between itself and user's head 208. To that end, mobile device 204 includes a sensor system (e.g., sensor system 120 of FIG. 1). The detected distance may be between the sensor system and a closest point on a user's head, a distance that is an average of distances to a portion of the user's head, or another distance. The sensor system, whether infrared, ultrasonic, or photographic, is capable of determining distance 206 and providing a corresponding representative electrical signal.

[0022] For example, if the sensor system is infrared, it may detect an infrared signal sent from mobile device 204 and reflected off of user's head 208. Using techniques known to those of skill in the art, such a reflected signal may be used to determine distance 206. Analogously, if ultrasonic, the sensor system may detect an ultrasonic signal transmitted from mobile device 204 and reflected off of user's head 208. Using techniques known to those of skill in the art, such a reflected signal may be used to determine distance 206. If photographic, the sensor system may use facial recognition logic to determine that user's head 208 is within a depth of field and, using techniques known to those of skill in the art, determine distance 206. Additionally if photographic information is acquired by an autofocus camera, distance 206 can be determined to be the focal distance of the camera's optical system. The autofocus system in this example can focus on the closest object, or on the specific region of the user's head, depending on the autofocus algorithm employed.

[0023] Any of the aforementioned techniques (infrared, ultrasonic, photographic) may be used in combination with acceleration data (e.g., detected by accelerometer 122) to calculate additional distances using, by way of non-limiting example, dead reckoning, known to those of skill in the art. For example, if an infrared, ultrasonic, or photographic technique is used to determine an absolute distance at a given time, and a subsequent acceleration in a direction away from the user's head is detected over a particular time interval, then, as known to those of skill in the art, these parameters are sufficient to derive an estimate of the absolute distance at the end (or during) the time interval. Regardless of the specific technology used to determine distance 206, mobile device 204 is capable of such determination.

[0024] Sensor systems (e.g., a photographic sensor) can also be used to determine a proportional change in distance by comparing the relative size of features on a user's head (e.g., an eye, an ear, a nose, or a mouth) and determining the proportional change in distance accordingly based on a reference size of the feature. In this way, the proportional change in distance can be used to perform the gain adjustments described herein without having to determine an absolute distance between the mobile device and the user.

[0025] FIG. 3 is a flow chart depicting a method of motion based compensation of downlinked audio according to various embodiments. In general, the perceived volume of audio emitted from a loudspeaker in a mobile device is a function of the distance between the mobile device loudspeaker and the listening user's ear(s). As the device gets further from the user's head, the perceived volume generally decreases. In general, doubling a distance from a sound source results in a decrease in perceived sound pressure of 6.02 dB. The method depicted in FIG. 3 may be used to compensate for perceived volume changes due to varying distance between a user's ear(s) and the loudspeaker emitting audio.

[0026] Thus, at block 300, a mobile device (e.g., mobile device 102 of FIG. 1 or 204 of FIG. 2) produces an electrical signal representing downlink audio. The electrical signal may be, by way of non-limiting example, an analog or digital signal representing the voice of a person with whom the user of the mobile device is communicating. Thus, the electrical signal may reflect information received from outside the device. In some embodiments, e.g., mobile devices that play pre-recorded music, the electrical signal may originate internal to the device.

[0027] At block 302, the distance between the device and the user's head is determined. As discussed above in reference to FIGs. 1 and 2, there are several techniques that may be employed to that end. For example, infrared distance detection or ultrasonic distance detection may be used. In general, mobile devices such as cellular telephones have a front face, which is generally pointed toward the user's head during operation. Accordingly, employing infrared or ultrasonic techniques to detect the distance to the nearest object before the front face of the mobile device may be implemented to achieve block 302. Alternately, or in addition, photographic facial recognition may be utilized. For such embodiments, the facial recognition techniques may detect the front of a person's face, or a person's face in profile and thereby determine the distance at issue. The aforementioned techniques may be used alone, in conjunction with one another, or in conjunction with a dead reckoning technique as informed by acceleration (e.g., using accelerometer 122 of FIG. 1) and timing information. Regardless as to the specific technique employed, block 302 results in the mobile device possessing data reflecting a distance from the device to the user's head.

[0028] At block 304, the gain level is set in accordance to the distance determined at block 302. In some embodiments, the gain level (e.g., gain of amplifier 112 of FIG. 1) is set in direct proportion to the distance measured. The table below reflects exemplary gain and sound pressure levels in relation to distance, where it is assumed by way of non-limiting example that, prior to any automatic adjustment according to the present embodiment, sound pressure at an initial distance of 1 cm from the source is 90 dB. Other proportionalities are also contemplated.

Distance	Uncompensated Sound Pressure Level	Gain
1 cm	90.00 dB	0.00 dB
2 cm	83.98 dB	6.02 dB
4 cm	77.96 dB	12.04 dB
8 cm	71.94 dB	18.06 dB
16 cm	65.92 dB	24.08 dB

Output Gain Table

In the above table, note that with each doubling of distance comes an additional 6.02 dB of gain used to compensate for the perceived decrease in volume.

[0029] At block 306, the audio is output from the loudspeaker. This may be achieved by feeding the output of a power amplifier directly to the loudspeaker (e.g., loudspeaker 114 of FIG. 1).

[0030] Flow from block 306 may return back to block 302 so that the gain is repeatedly adjusted. The repetitive adjustment may occur at periodic intervals (e.g., every 0.1 second, 0.5 second, or 1.0 second) as determined using a timer such as timer 124 of FIG. 1. Alternately, or in addition, the repetitive adjustment may be triggered by an event such as a detected acceleration of the device above a certain threshold.

[0031] Although an initial setting of 0 dB of gain for a distance of 1 cm is shown in the table above, the user may be more comfortable with another gain setting. Alternatively instead of an increase in gain as the distance is increased, the gain can be implemented as an increase in attenuation as distance is decreased. For example, in the case above, if the gain at 16 cm were to be 0 dB, the gain at 1 cm would then be -24.08 dB, or 24.08 dB of attenuation.

[0032] In addition, or in the alternative to the automatic adjustment of audio output gain, the audio input gain can also be adjusted as discussed below.

[0033] FIG. 4 is a flow chart depicting a method of motion based compensation of uplinked audio according to various embodiments. In general, the volume of audio picked up by a microphone varies with the distance between the microphone and the audio source. As the microphone gets farther away from the audio source, the amplitude of the detected sound decreases; as the microphone gets closer to the source, the amplitude of the detected sound increases. In general, doubling a distance between a sound source and microphone results in a decrease in sound pressure at the microphone of 6.02 dB. The method depicted in FIG. 4 may be used to compensate for sound pressure amplitude changes picked up by a microphone due to a varying distance between a user's mouth and a microphone of a mobile device.

[0034] Thus, at block 400, a mobile device (e.g., mobile device 102 of FIG. 1 or 204 of FIG. 2) receives sound at a microphone (e.g., microphone 118 of FIG. 1). At block 402, the sound is converted to an electrical signal. The electrical signal may be, by way of non-limiting example, an analog or digital signal representing the voice of user of the mobile device (including ambient noise).

[0035] At block 404, the distance between the device and the user's head is determined. As discussed above in reference to FIGs. 1 and 2, there are several techniques that may be employed to that end. For example, infrared distance detection or ultrasonic distance detection may be used. In general, mobile devices such as cellular telephones have a front face, which is generally pointed toward the user's head during operation. Accordingly, employing infrared or ultrasonic techniques to detect the distance to the nearest object before the front face of the mobile device may be implemented to achieve block 404. Alternately, or in addition, photographic facial recognition may be utilized. For such embodiments, the facial recognition techniques may detect the front of a person's face, or a person's face in profile and thereby determine the distance. Dead reckoning, as informed by acceleration information (e.g., gathered by accelerometer 122 of FIG. 1) may be performed in addition or in the alternative. Regardless as to the specific technique employed, block 404 results in the mobile device acquiring data reflecting a distance from the device to the user's head.

[0036] At block 406, the mobile device sets a gain of an amplifier of the electrical signal. In some embodiments, the gain level (e.g., gain of amplifier 116 of FIG. 1) is set in direct proportion to the distance determined at block 404. The amount of gain may compensate for the physical fact that as a distance between a user's mouth and the microphone increases, the detected sound at the microphone decreases. As discussed above, each doubling of distance results in a reduction of 6.02 dB of detected sound. Accordingly, the gain set at block 406 increases in a similar proportion. The following table illustrates an exemplary gain schedule, assuming a 0 dB gain in the amplifier when the user's mouth is a distance of 1 cm from the microphone.

Distance	Uncompensated Sound Pressure Level	Gain
1 cm	105.00 dB	0.00 dB
2 cm	98.98 dB	6.02 dB
4 cm	92.96 dB	12.04 dB
8 cm	86.94 dB	18.06 dB
16 cm	80.92 dB	24.08 dB

Input Gain Table

[0037] At block 408, audio filtering is modified to compensate for a so-called noise pumping effect. Specifically, if gain increases according to block 406, the noise within the captured audio also increases. Accordingly, if gain is increased by a certain number of decibels, a noise filter may be set to reduce noise by a corresponding or identical amount. The filter may be, by way of non-limiting example, a finite impulse response (FIR) filter set to filter noise at particular frequencies at which it occurs. Further details of a particular technique according to block 408 are discussed below in reference to FIG. 6.

[0038] At block 410, an output signal is generated. The output signal may be the result of the gain adjustment of block 406 and the noise reduction of block 408 applied to the electrical signal received at block 402. In some embodiments, the output signal is an analog signal to be stored in the mobile device; in other embodiments, the output signal is transmitted, e.g., to a cellular tower.

[0039] Flow from block 410 may return back to block 404 so that the gain may be repeatedly adjusted. The repetitive adjustment may occur at periodic intervals (e.g., every 0.1 second, 0.5 second, or 1.0 second) as determined using a timer such as timer 124 of FIG. 1. Alternately, or in addition, the repetitive adjustment may be triggered by an event such as a detected acceleration of the device above a certain threshold.

[0040] FIG. 5 is a flow chart depicting a method of intuitive motion based microphone gain adjustment according to various embodiments. Because not all users will speak at a similar sound level, a fixed input reference gain may not be applicable for all users. Due to this trait, an intuitive method of manually adjusting the input gain of a portable device is provided. In general, the technique illustrated by FIG. 5 allows a user to adjust a gain of a mobile device (e.g., mobile device 102 of FIG. 1) microphone using an intuitive, efficient, gesture-based procedure. The technique of FIG. 5 thus allows a user to set a gain for a microphone according to the user's preference. The gain adjusted may be that of a microphone on a cellular phone or other mobile computing device.

[0041] At block 500, the user provides a microphone gain set activation request to a mobile device. The microphone gain set activation request may be the user activating a physical or virtual (e.g., touchscreen) button on the mobile device. Alternately, or in addition, the microphone gain set activation request may be a voice command recognized by the device. The mobile device receives the request and enters a microphone gain adjustment mode, which the user controls as discussed presently. At block 502, the mobile device determines a distance to the user's head using any of the techniques disclosed herein (e.g., infrared, ultrasonic, or photographic, with or without dead reckoning).

[0042] At block 504, the mobile device adjusts an input gain for the microphone in inverse proportion to the distance. Thus, the farther the mobile device from the user's head, the more the gain level is lowered. Note that the microphone gain adjustment is made relative to the current gain set for the mobile device's microphone. Thus, for example, a user may hold the mobile device 10 cm from the user's head and request activation of the microphone gain set mode according to block 500. If the user brings the mobile device toward the user's head, the mobile device will increase the gain; if the user brings the mobile device away from the user's head, the mobile device will decrease the gain.

[0043] The proportionality of change in gain may be linear, quadratic, or another type of proportionality. For example, in some embodiments, each unit distance movement toward or away from the user's head (e.g., 1 cm) may result in an increase or decrease of gain by a fixed amount (e.g., 1 dB). As another example, in some embodiments, each unit distance movement toward or away from the user's head (e.g., 2 cm) may result in an increase or decrease of gain by an amount that is a function (e.g., a quadratic function) of the distance (e.g., $2^2 = 4$ dB). Exponential proportionalities are also contemplated. For example, each unit distance movement (e.g., x cm) may result in an increase or decrease of gain as an exponential function of the distance (e.g., 2^x dB).

[0044] Other embodiments may adjust microphone gain based on a change in relative distance. Thus, for example, some embodiments may use an initial distance from the user's head as a starting point. Each subsequent halving of the distance between the mobile device and the user's head may result in an increase of gain by a fixed amount (e.g., 6.02 dB), and each doubling of distance from the user's head may result in a decrease in gain by a fixed amount (e.g., 6.02 dB).

[0045] At block 506, the device provides input level feedback to the user. To provide user feedback during the adjustment process, one or more indicators can be displayed on the device informing the user of their speech level. A non-limiting example of such a feedback mechanism is a graphical (e.g., bar) indicator on the display of the device. The indicator could have acceptable reference input levels indicated on the display, allowing the user to adjust the input gain with the aforementioned motion compensation technique until the average speech falls within these bounds. In other embodiments, the feedback mechanism could be achieved through a change in color of an indicator, such as green (representing an acceptable level) and red (representing an unacceptable level). Further feedback mechanisms include a virtual sound level meter, or a non-visual indicator, such as tactile or audible feedback through the device (e.g., mechanical vibration or audible tones to warn of unacceptable levels).

[0046] At block 508, the device checks if it has received a microphone gain set inactivation request from the user. Reception of such a request causes the device to store 510 its gain level at its current state set during the operations of block 504. This stored value becomes the updated “anchor” for an updated input gain table. In some embodiments, the microphone gain set inactivation request may be the user activating a physical or virtual (e.g., touchscreen) button on the mobile device. In some embodiments, this may be the same button activated at block 500. The microphone gain set inactivation request may also be a voice command recognized by the device. If no activation request has been received, the flow returns to step 502 so that the gain can repeatedly be adjusted.

[0047] In other embodiments, when the microphone gain adjustment mode is activated, the adaptive gain control discussed in reference to FIG. 3 is disabled. In this case, as the distance between the device and the user’s head decreases, the received sound pressure level at the device naturally increases, and as the distance between the device and the user’s head increases, the received sound pressure level naturally decreases. Thus, step 504 does not change the gain electronically. When the received sound pressure level according to step 506 is acceptable to the user, the user initiates the microphone gain set inactivation request. The distance adaptive method of FIG. 4 is then reactivated using the current position as the reference gain level. The gain level will then be increased from this reference gain level as the device is moved further from the user's head, or decreased from this reference gain level as the device is moved closer to the user's head as shown in FIG. 4.

[0048] In some embodiments, the microphone gain set activation request of block 500 is made by activating and holding down a button (whether physical or virtual). In such embodiments, the microphone gain set inactivation request of block 508 may be made by releasing the same button. Thus, in such embodiments, the user employs the technique of FIG. 5 by initially holding the mobile device at a distance from the user's head, holding down an activation/deactivation button while adjusting the mobile device input gain by moving the mobile device toward or away from the user's head, and finally releasing the button after the user is satisfied with the resulting perceived microphone gain.

[0049] FIG. 6 is a flowchart depicting a method of noise abatement in uplinked audio according to various embodiments. The technique discussed in reference to FIG. 6 may be implemented, by way of non-limiting example, as part of block 408 of FIG. 4. In general, the technique discussed in reference to FIG. 6 serves to vary the amplitude in each frequency band of noise dynamically with the change in gain achieved at block 406 of FIG. 4 such that the overall signal-to-noise level is more consistent from time to time (or frame to frame, if frame-based signal processing is implemented). Thus, at block 600, a time period in which the user is not supplying sound to the microphone is identified. This may be performed, e.g., by setting a threshold and detecting when a detected sound level falls below the threshold or by using a voice activity detector (VAD) to detect when voice is not present in the microphone signal. The time period in which the user is not supplying sound is assumed to contain sound consisting mostly of noise.

[0050] At block 602, the frequency bands of the sound in association with block 600 are determined. This may be achieved using, for example, a Fourier transform or by dividing the audio spectrum into sub-bands. The frequency bands determined at block 602 represent the primary bands that contain the most noise. At block 604, audio filtering levels, or sub-band spectral suppression levels, are adjusted to reduce noise in the bands identified in block 602. The amount of reduction (or increase) may correspond with the amount of gain added (or reduced) at block 406 of FIG. 4.

[0051] Thus, for example, if a particular band identified as containing of mostly noise has a typical suppression value of, for example, 20 dB, and an additional 6 dB of gain is imposed at block 406 of FIG. 4 due to a user moving a mobile device away from the user's mouth, the noise suppression value for the filter at the particular band may be changed by a corresponding 6 dB, for a 26 dB suppression value. Likewise, if gain is reduced by 4 dB at block 406 of FIG. 4 due to a user moving the mobile device closer to the user's head, the suppression of the particular band may be set to 20 dB – 4 dB = 16 dB. This process may be performed for each noise band identified at block 602. The particular values presented herein are for illustration only and are not limiting.

[0052] The technique of FIG. 6 may be performed dynamically, periodically, or whenever a period of time in which no user sound is detected. Thus, the technique of FIG. 6 may be performed at block 408 of FIG. 4, but may also, or in the alternative, be performed at other times (e.g., at or between any of the blocks of FIG. 4).

[0053] FIG. 7 is a flowchart depicting a method of compensating for a Doppler effect in uplinked audio according to various embodiments. In general, if a user's mouth travels at a non-zero velocity relative to a microphone (e.g., microphone 118 of FIG. 1) while talking, the sound detected by such microphone will be pitch shifted according to the Doppler effect. The technique disclosed in reference to FIG. 7 may be used to compensate for such pitch shifting. In particular, the technique of FIG. 7 may be implemented together with the techniques discussed in any, or a combination, of FIGs. 3-6.

[0054] Thus, at block 700, a velocity of the mobile device (e.g., mobile device 102 of FIG. 1) relative to a user's head is determined. The techniques disclosed herein for determining a distance between a device and a user's head (infrared, ultrasonic, photographic, integration of acceleration) may be employed to determine velocity. More particularly, the disclosed distance-determining techniques may be repeated at short intervals (e.g., 0.01 seconds, 0.1 seconds) in order to detect changes in distance. Velocity may be calculated according to the changes in distance and corresponding time interval over which the distance changes are determined according to the formula $v=\Delta d/\Delta t$, where v represents velocity, Δd represents change in distance, and Δt represents change in time. Alternately, or in addition, information received from an accelerometer (e.g., accelerometer 122 of FIG. 1) may be used to determine relative velocity. Alternately, or in addition, the velocity can be taken directly from a velocity sensor contained in, e.g., sensor system 120 of FIG. 1.

[0055] Alternative techniques for determining device velocity can also be used when either distance or acceleration are sampled at a repetitive rate. For example if the distance or acceleration is sampled many times each second at a constant rate, a distance or acceleration time signal can be created. Because the velocity is the derivative of the distance time signal or the integral of the acceleration time signal, the velocity can be calculated in either the time or frequency domain. Suitable techniques include differentiating the distance signal in the time domain or integrating the acceleration signal in the time domain. An alternative technique is to convert the time signal into the frequency domain and either multiply each fast Fourier transform (FFT) bin value of the distance signal by the frequency of each FFT bin or divide each FFT bin value of the acceleration signal by the frequency of each FFT bin.

[0056] At block 702, the sound is adjusted to account for any Doppler shift caused by the velocity detected at block 700. In particular, the mobile device may include a look-up table or formula containing correspondences between velocity and pitch shift. After the velocity is determined at block 700, the corresponding pitch shift may be determined by such table or formula. The pitch shift may be adjusted in real-time using resampling technology to pitch shift or frequency scale, as is known in the art.

[0057] If direct velocity sensing, acceleration sensing, or proportional distance measurement are utilized, the Doppler shift compensation can be implemented without knowing the absolute distance between the mobile device and the user, just as the gain compensation can be implemented using only a proportional distance measure. In the cases of direct velocity sensing or acceleration sensing, this would not require any distance information to perform the Doppler shift. Thus the Doppler compensation can operate independent from a distance sensing operation.

[0058] In another embodiment, the method of compensating for a Doppler effect in FIG. 7 can be applied to downlink audio. As the loudspeaker in the device moves relative to the user's ears, a Doppler shift is present in the audio reaching the user's ears. The same methods of determining velocity for the uplink case (infrared, ultrasonic, photographic, velocity sensing, integration of acceleration data) can be used to determine velocity in the down link case. After the velocity of the device relative to the user's head is known, the audio being sent to the loudspeaker can be preprocessed using known pitch shifting techniques to adjust for the Doppler shift in the audio signal perceived by the user (e.g., after step 304 of FIG. 3).

[0059] In some embodiments, both the uplink and down link audio can be modified simultaneously to compensate for amplitude modulation as well as Doppler shift in the uplink and down link audio signals.

[0060] The foregoing description is illustrative, and variations in configuration and implementation may occur to persons skilled in the art. Other resources described as singular or integrated can in embodiments be plural or distributed, and resources described as multiple or distributed can in embodiments be combined. The scope of the present teachings is accordingly intended to be limited only by the following claims.

CLAIMS

What is claimed is:

1. A method of compensating for movement of a microphone relative to a user's head, wherein the microphone is present in a device, the method comprising:

producing, by the device, an electrical signal representative of audio received at the microphone;

determining, by the device, a distance between the device and the user's head;

setting, by the device, a gain of the electrical signal in accordance with the distance;

receiving an input gain set activation request;

ascertaining, by the device, a change in distance between the device and the user's head relative to the distance previously determined;

adjusting the gain of the electrical signal in inverse proportion to the change in distance between the device and the user's head;

receiving an input gain set inactivation request; and

generating, by the device, an output signal representative of the audio with the gain as adjusted.

2. The method of claim 1, further comprising:

modifying, by the device, an audio filtering in accordance with the change in distance, wherein the audio filtering is applied to the electrical signal;

wherein the generating further comprises generating, by the device, the output signal representative of the audio with the audio filtering

3. The method of claim 2, wherein the wherein the modifying comprises:

determining, by the device, a portion of the electrical signal corresponding to a time period when the user is not providing sound to the microphone;

determining frequency bands corresponding to audio in the portion of the electrical signal; and

adjusting the audio filtering to reduce audio in the frequency bands.

4. The method of claim 1, wherein the adjusting comprises:

increasing the gain of the electrical signal when the distance between the device and the user's head decreases; and

decreasing the gain of the electrical signal when the distance between the device and the user's head increases.

5. The method of claim 1, further comprising:

providing feedback to the user indicative of the gain of the microphone.

6. The method of claim 1, wherein the output signal comprises a cellular telephone signal.

7. The method of claim 1, further comprising:

determining, by the device, a velocity of the microphone relative to the user's head;

processing the electrical signal to compensate for a Doppler effect caused by the velocity.

8. The method of claim 1, wherein the device comprises a front of the device, and wherein the determining, by the device, the distance between the device and the user's head comprises:

determining, by the device, a distance between the front of the device and an object situated before the front of the device.

9. The method of claim 8, wherein the determining, by the device, the distance between the front of the device and the object situated before the front of the device comprises:

sending an infrared signal or an ultrasonic signal from the front of the device to the object.

10. The method of claim 1, wherein the determining, by the device, the distance between the device and the user's head comprises:

automatically detecting a human face.

11. An apparatus for compensating for movement of a microphone relative to a user's head, wherein the microphone is present in a device, the apparatus comprising:

a microphone configured to produce an electrical signal representative of audio received at the microphone;

a sensor system configured to determine a distance between the device and the user's head;

a user interface configured to receive an input gain set activation request;

logic, coupled to the user interface and the sensor system, configured to set a gain of the electrical signal in accordance with the distance and configured to adjust the gain of the microphone inversely proportional to a change in distance between the device and the user's head when the input gain set activation request is received;

an amplifier, coupled to the microphone and the logic; and

an output, operably coupled to the amplifier, configured to receive the electrical signal representative of the audio with the gain as adjusted by the logic.

12. The apparatus of claim 11, further comprising:

an audio filter configured for audio filtering in accordance with the distance, wherein the audio filtering is applied to the electrical signal;

wherein the output is further configured to generate the output signal representative of the audio with the audio filtering.

13. The apparatus of claim 12, further comprising:

logic configured to determine a portion of the electrical signal corresponding to a time period when the user is not providing sound to the microphone and also configured to determine frequency bands corresponding to audio in the portion of the electrical signal;

wherein the audio filter is further configured to reduce audio in the frequency bands.

14. The apparatus of claim 11, wherein the logic increases the gain of the electrical signal when the distance between the device and the user's head decreases and decreases the gain of the electrical signal when the distance between the device and the user's head increases

.

15. The apparatus of claim 11 configured to provide feedback to the user indicative of the gain of the microphone.

16. The apparatus of claim 11, wherein the output comprises a cellular telephone antenna.

17. The apparatus of claim 11, further comprising:

means for determining, by the device, a velocity of the microphone relative to the user's head; and

logic configured to process the electrical signal representative of audio to compensate for a Doppler effect caused by the velocity.

18. The apparatus of claim 11, wherein the sensor system comprises at least one of:

an infrared sensor, an ultrasonic sensor, or a camera.

19. The apparatus of claim 11, wherein the sensor system comprises:

an accelerometer.

20. The apparatus of claim 11, wherein the sensor system comprises:

a velocity sensor.

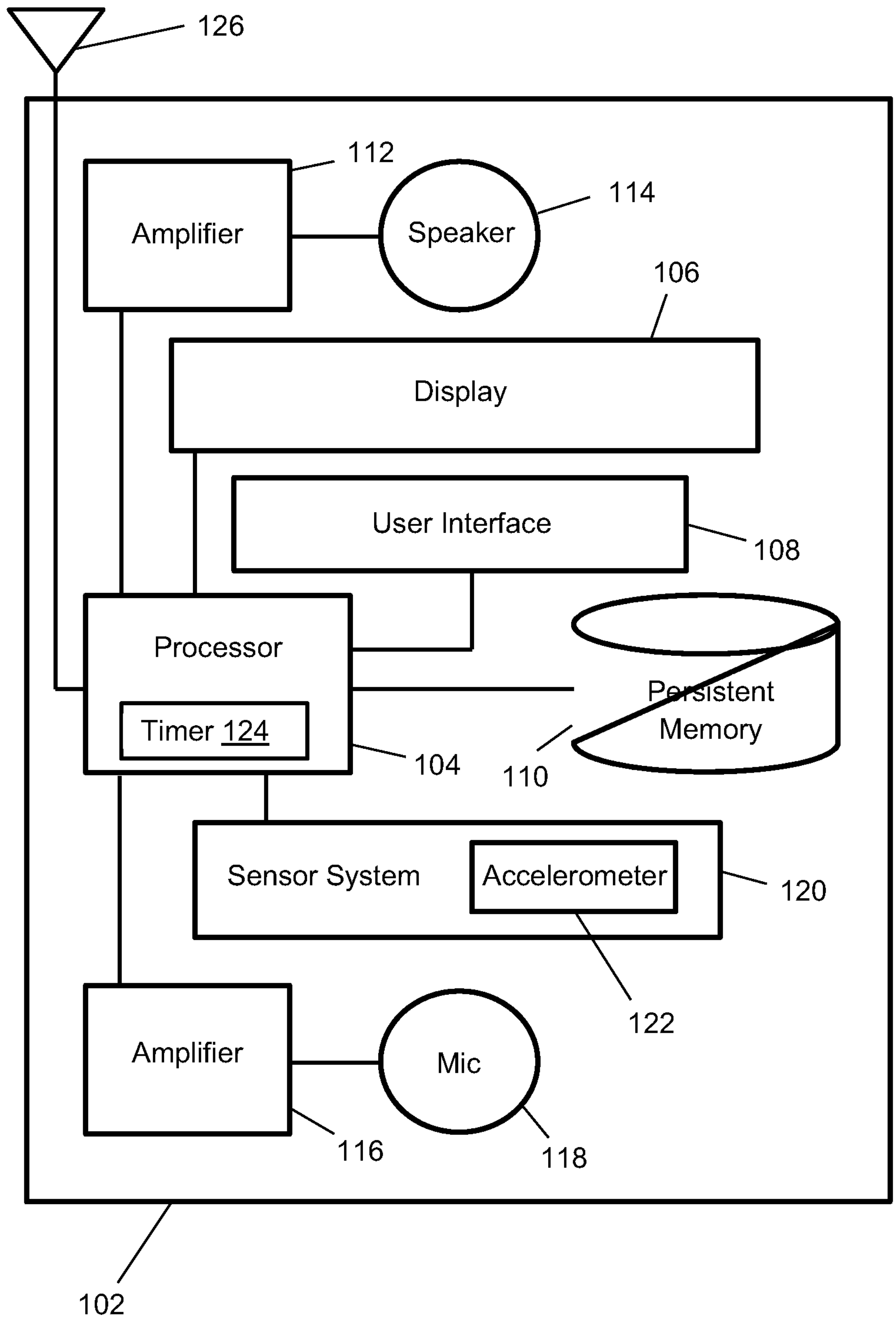
126
112
Amplifier
Speaker
114
106
Display
User Interface
108
Processor
Timer 124
Persistent Memory
104
110
Sensor System
Accelerometer
120
122
Amplifier
Mic
116
118
102

Fig. 1

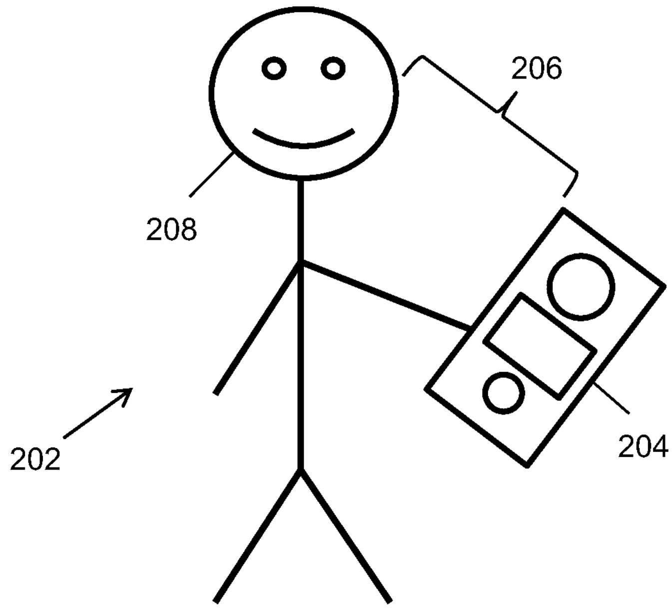

Fig. 2

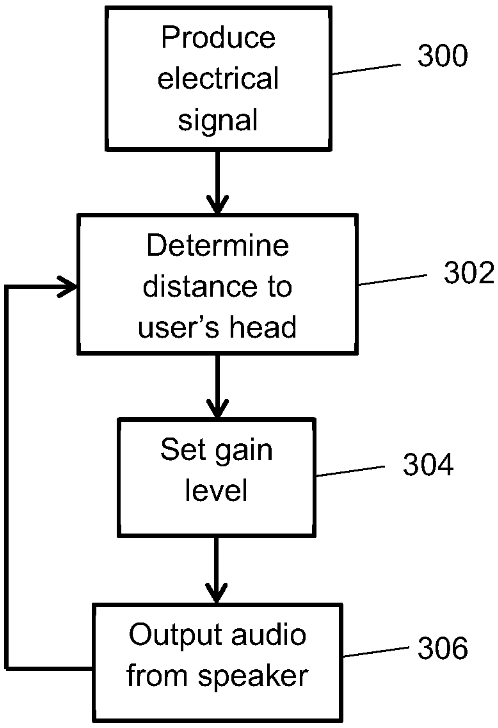
Produce electrical signal
300
Determine distance to user's head
302
Set gain level
304
Output audio from speaker
306

Fig. 3

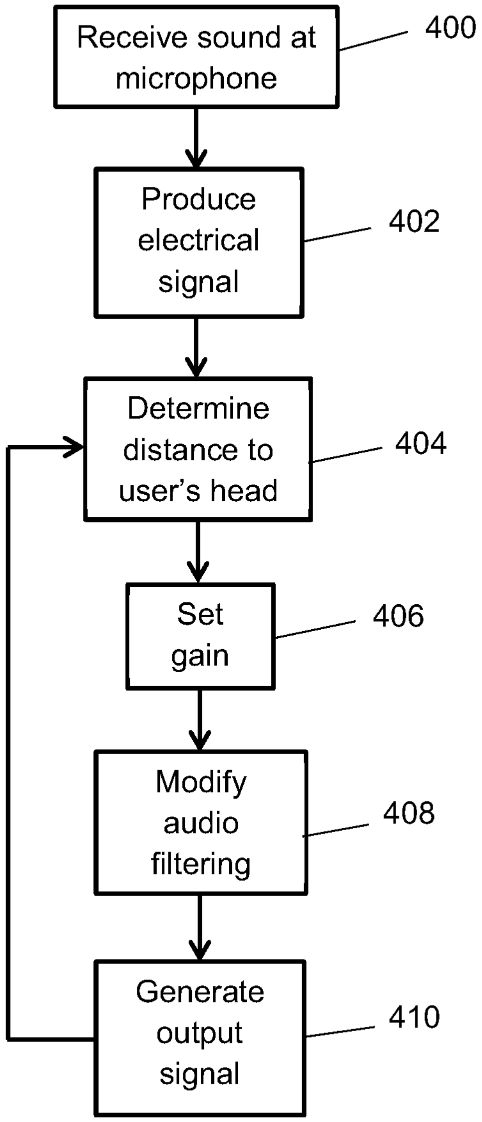
Receive sound at microphone
400
Produce electrical signal
402
Determine distance to user's head
404
Set gain
406
Modify audio filtering
408
Generate output signal
410

Fig. 4

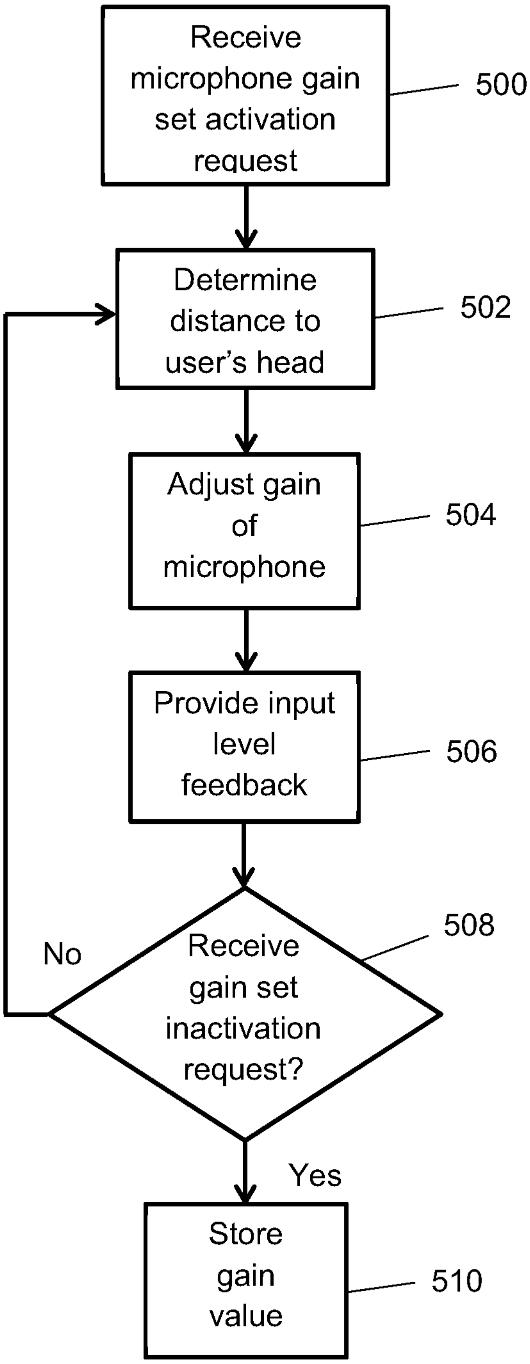
Receive microphone gain set activation request
500
Determine distance to user's head
502
Adjust gain of microphone
504
Provide input level feedback
506
Receive gain set inactivation request?
508
No
Yes
Store gain value
510

Fig. 5

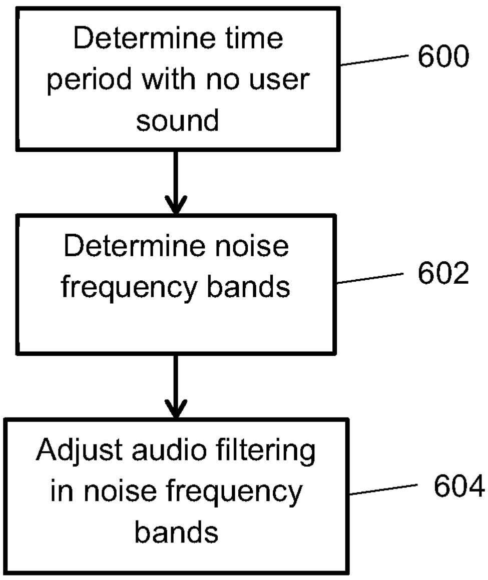
Determine time period with no user sound
600
Determine noise frequency bands
602
Adjust audio filtering in noise frequency bands
604

Fig. 6

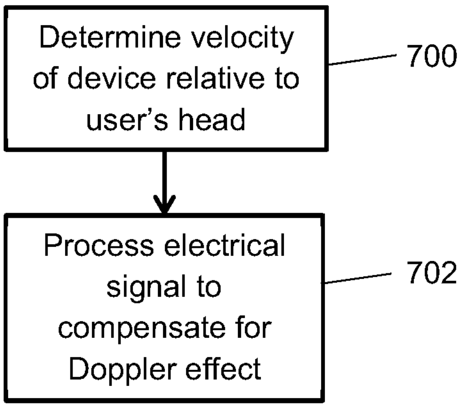
Determine velocity of device relative to user's head
700
Process electrical signal to compensate for Doppler effect
702

Fig. 7

INTERNATIONAL SEARCH REPORT

International application No
PCT/US2013/021552

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04M1/60 H03G3/32
ADD.

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)
H04M H03G

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)

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.
A	US 2007/202858 A1 (YU WEN-TING [TW]) 30 August 2007 (2007-08-30) paragraphs [0017], [0025], [0026]	1-20
A	US 2002/068537 A1 (SHIM JAE H [US] ET AL) 6 June 2002 (2002-06-06) paragraph [0005]	1-20

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

☒ See patent family annex.

* Special categories of cited documents :

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

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

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

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

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
11 April 2013	18/04/2013

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Fax: (+31-70) 340-3016	Amorotti, M

1

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No

PCT/US2013/021552

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 2007202858 A1	30-08-2007	JP 2007221744 A	30-08-2007
		US 2007202858 A1	30-08-2007
US 2002068537 A1	06-06-2002	NONE	