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(54) **Method for adjusting a system for providing hearing assistance to a user**

Methode zur Einstellung eines Hörhilfesystems

Méthode pour ajuster un système d'aide auditive

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(56) References cited:
EP-A- 1 443 803 WO-A-02/23948
DE-B3- 10 345 173 DE-B3-102004 025 691
US-A1- 2003 235 319

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Description

[0001] The present invention relates to a method for adjusting a system for providing hearing assistance to a user; it also relates to a corresponding system. In particular, the invention relates to a system comprising a microphone arrangement for capturing audio signals, a transmission unit for transmitting the audio signals via a wireless audio link from the transmission unit to a receiver unit, and means worn at or in the user's ear for stimulating the hearing of the user according to the audio signals received by the receiver unit.

[0002] Usually in such systems the wireless audio link is an FM radio link. According to a typical application of such wireless audio systems the receiver unit is connected to or integrated into a hearing instrument, such as a hearing aid, with the transmitted audio signals being mixed with audio signals captured by the microphone of the hearing instrument prior to being reproduced by the output transducer of the hearing instrument. The benefit of such systems is that the microphone of the hearing instrument can be supplemented or replaced by a remote microphone which produces audio signals which are transmitted wirelessly to the FM receiver and thus to the hearing instrument. In particular, FM systems have been standard equipment for children with hearing loss in educational settings for many years. Their merit lies in the fact that a microphone placed a few inches from the mouth of a person speaking receives speech at a much higher level than one placed several feet away. This increase in speech level corresponds to an increase in signal-to-noise ratio (SNR) due to the direct wireless connection to the listener's amplification system. The resulting improvements of signal level and SNR in the listener's ear are recognized as the primary benefits of FM radio systems, as hearing-impaired individuals are at a significant disadvantage when processing signals with a poor acoustical SNR.

[0003] Most FM systems in use today provide two or three different operating modes. The choices are to get the sound from: (1) the hearing instrument microphone alone, (2) the FM microphone alone, or (3) a combination of FM and hearing instrument microphones together.

[0004] Usually, most of the time the FM system is used in mode (3), i.e. the FM plus hearing instrument combination (often labeled "FM+M" or "FM+ENV" mode). This operating mode allows the listener to perceive the speaker's voice from the remote microphone with a good SNR while the integrated hearing instrument microphone allows the listener to also hear environmental sounds. This allows the user/listener to hear and monitor his own voice, as well as voices of other people or environmental noise, as long as the loudness balance between the FM signal and the signal coming from the hearing instrument microphone is properly adjusted. The so-called "FM advantage" measures the relative loudness of signals when both the FM signal and the hearing instrument microphone are active at the same time. As defined by the

ASHA (American Speech-Language-Hearing Association 2002), FM advantage compares the levels of the FM signal and the local microphone signal when the speaker and the user of an FM system are spaced by a distance of two meters. In this example, the voice of the speaker will travel 30 cm to the input of the FM microphone at a level of approximately 80 dB-SPL, whereas only about 65 dB-SPL will remain of this original signal after traveling the 2 m distance to the microphone in the hearing instrument. The ASHA guidelines recommend that the FM signal should have a level 10 dB higher than the level of the hearing instrument's microphone signal at the output of the user's hearing instrument.

[0005] When following the ASHA guidelines (or any similar recommendation), the relative gain, i.e. the ratio of the gain applied to the audio signals produced by the FM microphone and the gain applied to the audio signals produced by the hearing instrument microphone, has to be set to a fixed value in order to achieve e.g. the recommended FM advantage of 10dB under the above-mentioned specific conditions. Accordingly, - depending on the type of hearing instrument used - the audio output of the FM receiver has been adjusted in such a way that the desired FM advantage is either fixed or programmable by a professional, so that during use of the system the FM advantage - and hence the gain ratio - is constant in the FM+M mode of the FM receiver.

[0006] A method according to the preamble of claim 1 and a system according to the preamble of claim 25 are known from WO 02/23948 A1 which relates to an example of such an FM receiver which not only receives audio signals from a remote microphone transmitter but in addition may communicate with remote devices such as a remote control or a programming unit via wireless link for data transmission.

[0007] EP 1 638 367 A2 relates to another example of an FM receiver for receiving audio signals from a remote microphone transmitter, wherein the FM receiver upon receipt of a polling signal from the remote microphone transmitter is capable of transmitting status information regarding the FM receiver to the remote microphone transmitter.

[0008] WO 97/21325 A1 relates to a hearing system comprising a remote unit with a microphone and an FM transmitter and an FM receiver connected to a hearing aid equipped with a microphone. The hearing aid can be operated in three modes, i.e. "hearing aid only", "FM only" or "FM+M". In the FM+M mode the maximum loudness of the hearing aid microphone audio signal is reduced by a fixed value between and 10 dB below the maximum loudness of the FM microphone audio signal, for example by 4dB. Both the FM microphone and the hearing aid microphone may be provided with an automatic gain control (AGC) unit.

[0009] WO 02/30153 A1 relates to a hearing system comprising an FM receiver connected to a digital hearing aid, with the FM receiver comprising a digital output interface in order to increase the flexibility in signal treat-

ment compared to the usual audio input parallel to the hearing aid microphone, whereby the signal level can easily be individually adjusted to fit the microphone input and, if needed, different frequency characteristics can be applied. However, is not mentioned how such input adjustment can be done.

[0010] Contemporary digital hearing aids are capable of permanently performing a classification of the present auditory scene captured by the hearing aid microphones in order to select the hearing aid operation mode which is most appropriate for the determined present auditory scene. Examples for such hearing aids with auditory scene analyses can be found in US2002/0037087, US2002/0090098, WO 02/032208 and US2002/0150264.

[0011] DE 103 45 173 B3 relates to a remote control for a hearing aid which also serves to adjust the volume, i.e. the gain, of the hearing aid. The remote control may include a microphone for capturing audio signals in a region close to the remote control, which audio signals are transmitted to the hearing aid via a wireless link.

[0012] US 2003/0235319 relates to hearing aid connected by a wireless link to an external processor unit comprising a plurality of microphones and being designed for remaining long-term in a specific hearing environment.

[0013] EP 1 443 803 A2 relates to a hearing aid comprising means for the detection and automatic selection of an input signal from a plurality of sources.

[0014] DE 10 2004 025 691 B3 relates to a hearing aid comprising an acoustic classifier for adjusting the audio signal processing to the present acoustic situation.

[0015] Usually FM or inductive receivers are equipped with a squelch function by which the audio signal in the receiver is muted if the level of the demodulated audio signal is too low in order to avoid user's perception of excessive noise due a too low sound pressure level at the remote microphone or due to a large distance between the transmission unit and the receiver unit exceeding the reach of the FM link, see for example EP 0 671 818 B1 and EP 1 619 926 A1.

[0016] As already mentioned above, usually the FM advantage is set to a value of about 10 dB, which value is a compromise taking into account a medium surrounding noise level and a good intelligibility of both the FM audio signal and the voice of the neighbours. Further, this value is based on a medium sensitivity of the hearing aid audio input and on a specific microphone impedance of the hearing aid microphone. Variations of the audio input sensitivity of different hearing aids due to microphone impedance and/or sensitivity variations will have a direct impact on the desired FM advantage of 10 dB, i.e. they will cause a deviation from this desired value, resulting in a decreasing comprehension and listening comfort. Measurements have shown audio input sensitivity variations of up to ± 6 dB between the main hearing aid models present in the market. This implies that in practice the FM advantage will vary between 4 dB and

16 dB, depending on the hearing aid model connected to the FM receiver, instead of the desired value of 10 dB. In addition to that, tolerances of the FM transmitter and FM receiver gain are also added to the total FM advantage variation. Further, the desired FM advantage of 10 dB is a recommendation only and may not be optimum in any case or situation. In specific cases, the individual user's perception may require another value of the FM advantage than 10 dB.

[0017] It is an object of the invention to provide for a method for adjusting a system for providing hearing assistance to a user, wherein a remote microphone arrangement coupled by a wireless audio link to a receiver unit worn by the user is used and wherein perception of the transmitted audio signals should be optimized for the specific user, independently of the hearing instrument model and the FM system parameter variations and tolerances. It is a further object to provide for a corresponding system.

[0018] According to the invention, this object is achieved by a method as defined in claim 1 and by a system as defined in claim 25, respectively.

[0019] The invention is beneficial in that, by transmitting test audio signals to the receiver unit, simultaneously changing the gain by transmitting corresponding gain control commands to the receiver unit until an optimum value of the gain has been determined by the user, and storing that determined optimum gain value, undesired individual deviations of the perception of the audio signals from the remote microphone arrangement from the desired condition due to individual parameter variations and individual tolerances of the system can be avoided, so that for each practical individual system the desired optimum gain applied to the audio signals of the remote microphone arrangement can be determined and stored in order to use this optimum value during normal operation of the system.

[0020] Since the system comprises a hearing instrument which is worn at the user's ear and which is connected to the receiver unit or comprises the receiver unit, with the hearing instrument comprising the stimulating means, a second microphone arrangement for capturing second audio signals, and means for mixing the audio signals from the variable gain amplifier and the second audio signals prior to stimulating the user's hearing with the mixed audio signals via said stimulating means, the individual FM advantage, i.e. the ratio of the gain applied to the audio signals from the remote microphone arrangement applied to the audio signals from the hearing instrument microphone arrangement, can by be individually optimized regardless of individual parameter variations and individual tolerances.

[0021] Usually the audio signals from the receiver unit and the hearing instrument microphone will be mixed in the hearing instrument in such a manner that they are processed and power-amplified together so that gain applied to these audio signals in the hearing instrument is the same for both kinds of audio signals; consequently,

after mixing the gain ratio will not be changed by the usual dynamic audio signal processing of the hearing instrument. Thus, by controlling the gain applied to the audio signals from the remote microphone arrangement by the variable gain amplifier of the receiver unit, also the gain ratio, i.e. the ratio of the gain applied to the audio signals from the remote microphone arrangement and the gain applied to the audio signals from the hearing instrument microphone, can be controlled.

[0022] The parameter variations and tolerances which can be compensated by the adjustment method of the present invention include the following: microphone sensitivity of the radio transmitter, modulation strength of the radio transmitter, audio output level of the radio receiver, output impedance of the radio receiver, audio input sensitivity of the hearing aid, audio input impedance of the hearing aid, and specific sensitivity of the user.

[0023] According to one embodiment, the test audio signals are generated by retrieving audio signals from a memory. According to another embodiment, the test audio signals may be generated by an audio signal synthesiser. According to a further alternative embodiment, the test audio signals may be generated by generating a test sound which is captured as the test audio signals by the remote microphone arrangement; usually the test sound will be the voice of a person using the transmitting unit, such as a teacher. In this case, the test sound may be captured also by the second microphone arrangement, so that for optimizing the gain, and also the gain ratio, also the audio signals captured by the second microphone arrangement may be taken into account. Typically the test audio signal transmitted to the receiver unit will be transmitted at a maximum level of the audio signals of the remote microphone arrangement, which is typical when the person using the transmitting unit is speaking.

[0024] A data link for transmitting the commands to the receiver unit and the audio signal link may be realized by a common transmission channel, with the bandwidth being split.

[0025] According to one embodiment, the system may be operated in such a manner that the gain is kept constant at a value corresponding to the determined optimum value. According to an alternative embodiment, the system may be operated in such a manner that the gain is dynamically changed according to the result of a permanently repeated auditory scene analysis based on at least one of the audio signals provided by the remote microphone arrangement and the audio signals provided by the hearing instrument microphone arrangement. In this case the determined optimum value of the gain is used to calibrate the variable gain amplifier, i.e. the gain control algorithm is calibrated by the determined optimum gain value.

[0026] Preferred embodiments of the invention are defined in the dependent claims.

[0027] In the following, examples of the invention are described and illustrated by reference to the attached drawings, wherein:

Fig. 1 is a schematic view of the use of an embodiment of a hearing assistance system according to the invention;

5 Fig. 2 is a schematic view of the transmission unit of the system of Fig. 1;

Fig. 3 is a diagram showing the signal amplitude versus frequency of the common audio signal / data transmission channel of the system of Fig. 1;

Fig. 4 is a block diagram of one embodiment of the receiver unit of the system of Fig. 1;

15 Fig. 5 is a block diagram of one embodiment of the transmission unit of the system of Fig. 1;

Fig. 6 is a block diagram of another embodiment of the transmission unit of the system of Fig. 1;

20 Fig. 7 is a diagram showing an example of the gain set by the variable gain amplifier versus time;

Fig. 8 shows schematically an example in which the receiver unit is connected to a separate audio input of a hearing aid; and

Fig. 9 shows schematically an example in which the receiver unit is connected in parallel to the microphone arrangement of a hearing aid.

[0028] Fig. 1 shows schematically the use of a system for hearing assistance comprising an FM radio transmission unit 102 comprising a directional microphone arrangement 26 consisting of two omnidirectional microphones M 1 and M2 which are spaced apart by a distance d, an FM radio receiver unit 103, and hearing instrument 104 comprising a microphone arrangement 36. The audio output of the receiver unit 103 is connected to an audio input of the hearing instrument 104 via an audio shoe (not shown). The transmission unit 102 is worn by a speaker 100 around his neck by a neck-loop 121 acting as an FM radio antenna, with the microphone arrangement 26 capturing the sound waves 105 carrying the speaker's voice. Audio signals and control data are sent from the transmission unit 102 via radio link 107 to the receiver unit 103 worn by a user/listener 101. In addition to the voice 105 of the speaker 100 background/surrounding noise 106 may be present which will be both captured by the microphone arrangement 26 of the transmission unit 102 and microphone arrangement 36 of the hearing instrument 104. Typically the speaker 100 will be a teacher and the user 101 will be a hearing-impaired person in a classroom, with background noise 106 being generated by other pupils.

[0029] Fig. 8 is a block diagram of an example in which the receiver unit 103 is connected to a high impedance audio input of the hearing instrument 104. The receiver

unit 103 contains a module 31 for demodulation and signal processing for processing the FM signal received by the antenna 123 from the antenna of the transmission unit 102 (these audio signals resulting from the microphone arrangement 26 of the transmission unit 102 in the following also will be referred to as "first audio signals"). The processed first audio signals are amplified by variable gain amplifier 126. The output of the receiver unit 103 is connected to an audio input of the hearing instrument 104 which is separate from the microphone 36 of the hearing instrument 15 (such separate audio input has a high input impedance).

[0030] The first audio signals provided at the separate audio input of the hearing instrument 104 may undergo pre-amplification in a pre-amplifier 33, while the audio signals produced by the microphone 36 of the hearing instrument 104 (in the following referred to "second audio signals") may undergo pre-amplification in a pre-amplifier 37. The hearing instrument 104 further comprises a digital central unit 35 into which the first and second audio signals are supplied as a mixed audio signal for further audio signal processing and amplification prior to being supplied to the input of the output transducer 38 of the hearing instrument 104. The output transducer 38 serves to stimulate the user's hearing 39 according to the combined audio signals provided by the central unit 35.

[0031] Fig. 9 shows a modification of the embodiment of Fig. 8, wherein the output of the receiver unit 103 is not provided to a separate high impedance audio input of the hearing instrument 104 but rather is provided to an audio input of the hearing instrument 104 which is connected in parallel to the hearing instrument microphone 36. Also in this case, the first and second audio signals from the remote microphone arrangement 26 and the hearing instrument microphone 36, respectively, are provided as a combined/mixed audio signal to the central unit 35 of the hearing instrument 104. The gain applied to first audio signals can be adjusted by the variable gain amplifier 126 of the receiver unit 103. Further, also the gain ratio for the first and second audio signals can be controlled by the receiver unit 103 by accordingly controlling the signal at the audio output of the receiver unit 103 and the output impedance Z_1 of the audio output of the receiver unit 103.

[0032] Fig. 2 is a schematic view of the transmission unit 102 which, in addition to the microphone arrangement 26, comprises a digital signal processor 122, an FM transmitter 120, an antenna 149 for establishing a short distance bidirectional inductive link 54 with an antenna 151 of the receiver unit 103, a button 50 for activating an FM advantage adjustment mode of the transmission unit 102 and the receiver unit 103, a button 51 to read identification information stored in the receiver unit 103 via the inductive link 54, a button 52 for causing a "volume up" command being transmitted to the receiver unit 103, and a button 53 for causing a "volume down" command being transmitted to the receiver unit 103.

[0033] According to Fig. 3, the channel bandwidth of

the FM radio transmitter, which, for example, may range from 100 Hz to 7 kHz, is split in two parts ranging, for example from 100 Hz to 5 kHz and from 5 kHz to 7 kHz, respectively. In this case, the lower part is used to transmit the audio signals (i.e. the first audio signals) resulting from the microphone arrangement 26, while the upper part is used for transmitting data from the FM transmitter 120 to the receiver unit 103. The data link established thereby can be used for transmitting control commands relating to the gain from the transmission unit 102 to the receiver 103, and it also can be used for transmitting general information or commands to the receiver unit 103.

[0034] The internal architecture of the FM transmission unit 102 is schematically shown in Fig. 5. As already mentioned above, the spaced apart omnidirectional microphones M1 and M2 of the microphone arrangement 26 capture both the speaker's voice 105 and the surrounding noise 106 and produce corresponding audio signals which are converted into digital signals by the analog-to-digital converters 109 and 110. M1 is the front microphone and M2 is the rear microphone. The microphones M1 and M2 together associated to a beamformer algorithm form a directional microphone arrangement 26 which, according to Fig. 1, is placed at a relatively short distance to the mouth of the speaker 100 in order to insure a good SNR at the audio source and also to allow the use of easy to implement and fast algorithms for voice detection as will be explained in the following. The converted digital signals from the microphones M1 and M2 are supplied to the unit 111 which comprises a beamformer implemented by a classical beamformer algorithm and a 5 kHz low pass filter. The first audio signals leaving the beamformer unit 111 are supplied to a gain model unit 112 which mainly consists of an automatic gain control (AGC) for avoiding an overmodulation of the transmitted audio signals. The output of a gain model unit 112 is supplied to an adder unit 113 which mixes the first audio signals, which are limited to a range of 100 Hz to 5 kHz due to the 5 kHz low pass filter in the unit 111, and DTMF (dual-tone multi-frequency) encoded data signals supplied from a control unit 162 within a range from 5 kHz and 7 kHz. The combined audio/data signals are converted to analog by a digital-to-analog converter 119 and then are supplied to the FM transmitter 120 which uses the neck-loop 121 as an FM radio antenna.

[0035] The transmission unit 102 further comprises a voice memory 160 in which test audio signals are stored which can be retrieved by request of a control unit 162 and which are then supplied to the gain model unit 112. The control unit 162 generates commands for controlling the transmission unit 102 and the receiver unit 103 according to operation of the buttons 50 to 53 by the user 100. Such control commands are transmitted via the FM transmitter 120 and the antenna 121 to the receiver unit 103. The units 109, 110, 111, 112, 113, 119 and 162 all can be realized by the digital signal processor 122 of the transmission unit 102.

[0036] The receiver unit 103 is schematically shown in Fig. 4. The audio signals produced by the microphone arrangement 26 and processed by the units 111 and 112 of transmission unit 102 and the command signals produced by the control unit 162 of the transmission unit 102 are transmitted from the transmission unit 102 over the same FM radio channel to the receiver unit 103 where the FM radio signals are received by the antenna 123 and are demodulated in an FM radio receiver 124. An audio signal low pass filter 125 operating at 5 kHz supplies the audio signals to a variable gain amplifier 126 from where the audio signals are supplied to the audio input of the hearing instrument 104. The output signal of the FM radio receiver 124 is also filtered by a high pass filter 127 operating at 5 kHz in order to extract the commands from the control unit 162 contained in the FM radio signal. A filtered signal is supplied to a unit 128 including a DTMF and digital demodulator/decoder in order to decode the command signals from the control unit 162.

[0037] The command signals decoded in the unit 128 are provided to a parameter update unit 129 in which the parameters of the commands are updated according to information stored in an EEPROM 130 of the receiver unit 103. The output of the parameter update unit 129 is used to control the variable gain amplifier 126 which is gain and output impedance controlled. Thereby the audio signal output of the receiver unit 103 can be controlled according to the commands from the control unit 162 in order to control the gain (and also the gain ratio, i.e. the ratio of the gain applied to the audio signals from the microphone arrangement 26 of the transmission unit 102 and the audio signals from the hearing instrument microphone 36) according to the commands from the control unit 162.

[0038] The inductive antenna 151 of the receiver unit 103 is connected via a unit 150 to the EEPROM 130 and is used for reading identification information stored in the EEPROM 130, which serves to identify the receiver unit 103, via the inductive link 54 by the transmission unit 102. In addition, the inductive link 54 may have additional functions such as reading other receiver parameters, programming the receiver unit 103, monitoring battery status, the receiver unit 103 and monitoring the quality of the link.

[0039] The desired gain determined by the amplifier 126 may be adjusted according to the following procedure.

[0040] First, the user 100 selects the respective receiver unit 103, which is to be adjusted by approaching the receiver unit 103 with the transmission unit 102 so close that the receiver unit 103 comes within the reach of the inductive link 54. Then the button 51 is pushed whereby the control unit 162 causes the transmission unit 102 to read the identification code via the inductive link 54 from the EEPROM 130 of the receiver unit 103. Once the identification code has been read by the transmission unit 102, this particular identification code is coded over the data link of the transmission unit 102 in order to address

in the further adjustment procedure only the specified receiver unit 103. If the user 101 uses two hearing instruments 104, two receiver units 103 must be addressed by the transmission unit 102. If the user 101 is the only one within the reach distance of the transmission unit 102, the receiver identification step can be omitted.

[0041] As a next step, the user 100 will enter an adjustment mode of the transmission unit 102 by pushing the button 50.

[0042] In the FM advantage adjustment procedure then test audio signal is generated, for example, by retrieving a test signal from the voice memory 160. Alternatively, the test audio signals may be generated by the voice of the user 100 which is captured by the microphone arrangement 26. In the latter case, the voice of the user 100 also will be captured by the hearing instrument microphone 36. In any case, the test audio signal preferably will be transmitted to the receiver unit 103 at the maximum audio level of the transmission unit 102, which is typical for the case when the user 100 is speaking. The test audio signals provided by the low pass filter 125 will be amplified by the amplifier 126 according to the presently set gain in the EEPROM 130 and then will be supplied to the hearing instrument 104 for being reproduced by the speaker 38.

[0043] As a next step, perception of the test audio signals by the user 101 will be evaluated, and according to the result of this evaluation the volume-up-button 52 will be pushed if the user 101 feels that the volume of the audio test signals is too low, or the volume-down-button 53 will be pushed if the user 101 feels that the volume of the test audio signals is too high. Upon operation of the respective button 52 or 53 the control unit 162 will cause a corresponding control command to be transmitted to the receiver unit 103 where it is demodulated in the unit 128 and serves to correspondingly increase or reduce the gain applied by the amplifier 126 via the unit 129.

[0044] Such change of the gain applied by the amplifier 126 is continued until an optimum value - which corresponds then to the optimum value of the individual FM advantage - has been found. Thereupon that determined optimum gain value will be stored in the EEPROM 130 of the receiver unit upon receipt of a respective command sent by the transmitting unit 102. Such store command signal may be generated by the control unit 162 of the transmission unit 102 upon corresponding operation of the buttons at the transmission unit 102, for example by again pushing the "A"-button 50, or it may be generated automatically, if a certain time period without operation of the volume up or volume down-buttons 52, 53 has lapsed.

[0045] After having terminated the FM advantage adjustment procedure, the transmission unit 102 and the receiver unit 103 will resume the normal operation mode. This normal operation mode may be such that the determined optimum gain value stored in the EEPROM 130 will be continuously applied to the amplifier 126, i.e. the amplifier 126 will be operated at constant gain.

[0046] According to an alternative embodiment which is shown in Figs. 6 and 7, the transmission unit 102 and the receiver unit 103 may be designed such that in the normal operation mode the gain presently applied by the amplifier 126 may be changed according to the result of an auditory scene analysis permanently performed by the transmission unit 102 by analysing the audio signal captured by the microphone arrangement 26. The receiver unit 103 shown in Fig. 4 may be used also with the transmission unit 102 of Fig. 6.

[0047] To this end, the transmission unit 102 is provided with classification unit 134, the functions of which may be implemented by the digital signal processor 122. The classification unit 134 shown in Fig. 6 includes units 114, 115, 116, 117 and 118, as will be explained in detail in the following.

[0048] The unit 114 is a voice energy estimator unit which uses the output signal of the beam former unit 111 in order to compute the total energy contained in the voice spectrum with a fast attack time in the range of a few milliseconds, preferably not more than 10 milliseconds. By using such short attack time it is ensured that the system is able to react very fast when the speaker 11 begins to speak. The output of the voice energy estimator unit 114 is provided to a voice judgement unit 115 which decides, depending on the signal provided by the voice energy estimator 114, whether close voice, i.e. the speaker's voice, is present at the microphone arrangement 26 or not.

[0049] The unit 117 is a surrounding noise level estimator unit which uses the audio signal produced by the omnidirectional rear microphone M2 in order to estimate the surrounding noise level present at the microphone arrangement 26. However, it can be assumed that the surrounding noise level estimated at the microphone arrangement 26 is a good indication also for the surrounding noise level present at the microphone 36 of the hearing instrument 104, like in classrooms for example. The surrounding noise level estimator unit 117 is active only if no close voice is presently detected by the voice judgement unit 115 (in case that close voice is detected by the voice judgement unit 115, the surrounding noise level estimator unit 117 is disabled by a corresponding signal from the voice judgment unit 115). A very long time constant in the range of 10 seconds is applied by the surrounding noise level estimator unit 117. The surrounding noise level estimator unit 117 measures and analyzes the total energy contained in the whole spectrum of the audio signal of the microphone M2 (usually the surrounding noise in a classroom is caused by the voices of other pupils in the classroom). The long time constant ensures that only the time-averaged surrounding noise is measured and analyzed, but not specific short noise events. According to the level estimated by the unit 117, a hysteresis function and a level definition is then applied in the level definition unit 118, and the data provided by the level definition unit 118 is supplied to the unit 116 in which the data is encoded by a digital encoder/modulator and

is transmitted continuously with a digital modulation having a spectrum a range between 5 kHz and 7 kHz. That kind of modulation allows only relatively low bit rates and is well adapted for transmitting slowly varying parameters like the surrounding noise level provided by the level definition unit 118.

[0050] The estimated surrounding noise level definition provided by the level definition unit 118 is also supplied to the voice judgement unit 115 in order to be used to adapt accordingly to it the threshold level for the close voice/no close voice decision made by the voice judgement unit 115 in order to maintain a good SNR for the voice detection.

[0051] If close voice is detected by the voice judgement unit 115, a very fast DTMF (dual-tone multi-frequency) command is generated by a DTMF generator included in the unit 116. The DTMF generator uses frequencies in the range of 5 kHz to 7 kHz. The benefit of such DTMF modulation is that the generation and the decoding of the commands are very fast, in the range of a few milliseconds. This feature is very important for being able to send a very fast "voice ON" command to the receiver unit 103 in order to catch the beginning of a sentence spoken by the speaker 11. The command signals produced in the unit 116 (i.e. DTMF tones and continuous digital modulation) are provided to the adder unit 113, as already mentioned above.

[0052] Fig. 7 illustrates an example of how the gain in the normal operation mode may be controlled according to the determined present auditory scene category.

[0053] As already explained above, the voice judgement unit 115 provides at its output for a parameter signal which may have two different values:

(a) "Voice ON": This value is provided at the output if the voice judgement unit 115 has decided that close voice is present at the microphone arrangement 26. In this case, fast DTMF modulation occurs in the unit 116 and a control command is issued by the unit 116 and is transmitted to the amplifier 126, according to which the gain is set to a given value which, for example, may result in an FM advantage of 10 dB under the respective conditions of, for example, the ASHA guidelines.

(b) "Voice OFF": If the voice judgement unit 115 decides that no more close voice is present at the microphone arrangement 26, a "voice OFF" command is issued by the unit 116 and is transmitted to the amplifier 126. In this case, the parameter update unit 129 applies a "hold on time" constant 131 and then a "release time" constant 132 defined in the EEPROM 130 to the amplifier 126. During the "hold on time" the gain set by the amplifier 126 remains at the value applied during "voice ON". During the "release time" the gain set by the amplifier 126 is progressively reduced from the value applied during "voice ON" to a lower value corresponding to a "pause at-

tenuation" value 133 stored in the EEPROM 130. Hence, in case of "voice OFF" the gain of the microphone arrangement 26 is reduced relative to the gain of the hearing instrument microphone 36 compared to "voice ON". This ensures an optimum SNR for the hearing instrument microphone 36, since at that time no useful audio signal is present at the microphone arrangement 26 of the transmission unit 102.

[0054] The control data/command issued by the surrounding noise level definition unit 118 is the "surrounding noise level" which has a value according to the detected surrounding noise level. As already mentioned above, the "surrounding noise level" is estimated only during "voice OFF" but the level values are sent continuously over the data link. Depending on the "surrounding noise level" the parameter update unit 129 controls the amplifier 126 such that according to definition stored in the EEPROM 130 the amplifier 126 applies an additional gain offset or an output impedance change to the audio output of the receiver unit 103.

[0055] The application of an additional gain offset is preferred in case that there is the relatively low surrounding noise level (i.e. quiet environment), with the gain of the hearing instrument microphone 36 being kept constant. The change of the output impedance is preferred in case that there is a relatively high surrounding noise level (noisy environment), with the signals from the hearing instrument microphone 36 being attenuated by a corresponding output impedance change. In both cases, a constant SNR for the signal of the microphone arrangement 26 compared to the signal of the hearing instrument microphone 36 is ensured.

[0056] A preferred application of the systems according to the invention is teaching of pupils with hearing loss in a classroom. In this case the speaker 100 is the teacher, while a user 101 is one of several pupils, with the hearing instrument 104 being a hearing aid.

[0057] The FM advantage adjustment procedure in the adjustment mode may be similar to that described above with regard to the system of Figs. 4 and 5. In the case of the embodiment of Figs. 6 and 7 the optimum gain value determined and stored in the adjustment mode will be used to calibrate the gain variation based on the auditory scene analysis in the normal operation mode. In present case, for example, the value of the gain applied in the "Voice ON" regime will correspond to the optimum gain value determined and stored in the adjustment mode.

[0058] While in the embodiments described so far the receiver unit is separate from the hearing instrument, in some embodiments it may be integrated with the hearing instrument.

[0059] The microphone arrangement producing the second audio signals may be connected to or integrated within the hearing instrument. The second audio signals may undergo an automatic gain control prior to being mixed with the first audio signals. The microphone ar-

rangement producing the second audio signals may be designed as a directional microphone comprising two spaced apart microphones.

Claims

1. A method for adjusting a system for providing hearing assistance to a user (101), the system comprising a microphone arrangement (26) for capturing audio signals, a transmission unit (102) for transmitting the audio signals via a wireless link (107) to a receiver unit (103) worn by the user, a variable gain amplifier (126) located in the receiver unit for applying a gain to the audio signals, and a hearing instrument (104) which is worn at the user's ear (39) and which is connected to the receiver unit (103) or comprises the receiver unit, said hearing instrument comprising means (38) worn at or in the user's ear (39) for stimulating the hearing of the user according to the audio signals from the variable gain amplifier (126), a second microphone arrangement (36) for capturing second audio signals, and means for mixing the audio signals from the variable gain amplifier (126) and the second audio signals prior to stimulating the user's hearing with the mixed audio signals via said stimulating means **characterized by:**

- (a) generating test audio signals, transmitting said test audio signals at a pre-defined level from the transmission unit via the wireless link to the receiver unit and stimulating the user's hearing with said test audio signals via said stimulating means;
- (b) simultaneously transmitting gain control commands from the transmission unit to the variable gain amplifier in order to selectively change the gain applied by the variable gain amplifier (126);
- (c) repeating steps (a) and (b) until an optimum value of the gain is applied by the variable gain amplifier (126); and
- (d) transmitting a store command from the transmission unit to the receiver unit in order to store that determined optimum value of the gain.

2. The method of claim 1, wherein in step (a) said test audio signals are generated by retrieving audio signals from an audio signal memory (160).

3. The method of claim 2, wherein said audio signal memory (160) is integrated in the transmission unit (102).

4. The method of claim 1, wherein in step (a) said test audio signals generated by an audio signal synthesizer.

5. The method of claim 4, wherein said audio signal synthesizer is integrated within the transmission unit (102).
6. The method of claim 1, wherein in step (a) said test audio signals are generated by generating a test sound and capturing said test sound as said test audio signals by the microphone arrangement (26).
7. The method of claim 6, wherein said test sound is the voice of a person (100) using the transmission unit (102).
8. The method of one of claims 6 and 7, further comprising: capturing said test sound as said second audio signals by the second microphone arrangement (36), mixing the audio signals from the variable gain amplifier (126) and the second audio signals according to the presently set gain and stimulating the user's hearing with said mixed audio signals via said stimulating means of the hearing instrument
9. The method of one of the preceding claims, wherein in step (d) the determined optimum value of the gain is stored in a memory (130) which is integrated within the receiver unit (103) or the hearing instrument (104).
10. The method of claim 9, wherein prior to step (a) the receiver unit (103) is identified by reading an identification information stored in the receiver unit by the transmission unit (102) via an inductive link (54).
11. The method of claim 10, wherein the receiver unit (103) is specifically addressed by the transmission unit (102) by transmitting a signal coded according to the identification information read by the transmission unit.
12. The method of one of the preceding claims, wherein in step (a) the test signal is transmitted at a maximum level of the audio signals of the transmission unit (102).
13. The method of claim 1, wherein the receiver unit (103) is connected to the hearing instrument (104) and the variable gain amplifier (126) is gain and/or output impedance controlled.
14. The method of one of the preceding claims, wherein a data link for transmitting the gain control commands and the store command and the audio signal link (106) are realized by a common transmission channel.
15. The method of claim 14, wherein the lower portion of the bandwidth of the transmission channel is used by the audio signal link (106) and the upper portion of the bandwidth of the channel is used by the data link.
16. The method of claim 1, wherein the output of the receiver unit (103) is connected in parallel with the second microphone arrangement (36).
17. The method of claim 1, wherein the audio signals from the receiver unit (103) are supplied to the hearing instrument (104) via an audio input separate from the second microphone arrangement (36).
18. The method of one of the preceding claims, wherein the audio signal link is an FM radio link
19. The method of claim 1, wherein the hearing instrument (104) is a hearing aid having an electroacoustic output transducer (38) as the stimulating means.
20. The method of one of the preceding claims, wherein the audio signals in the transmission unit (102) undergo an automatic gain control treatment in a gain model unit (112) prior to being transmitted to the receiver unit.
21. The method of one of the preceding claims, wherein the gain applied by the variable gain amplifier (126) is a constant value, with said constant value corresponding to the stored optimum value of the gain.
22. The method of one of claims 1 to 21, further comprising
 - (a) capturing audio signals by the microphone arrangement (26) and transmitting the audio signals by the transmission unit (102) via the wireless audio link (107) to the receiver unit (103);
 - (b) analyzing the audio signals prior to being transmitted by a classification unit (134) in order to determine a present auditory scene category from a plurality of auditory scene categories;
 - (c) applying a gain by the variable gain amplifier (126) to the audio signals, which gain is selected according to the present auditory scene category determined in step (b);
 - (d) stimulating the user's hearing by the stimulating means (38) according to the audio signals from the variable gain amplifier (126);
- wherein the stored optimum value of the gain is used to calibrate the variable gain amplifier (126).
23. The method of claim 22, wherein the gain applied for at least onto of the auditory scenes is the stored optimum value of the gain.
24. The method of claim 23, wherein the gain applied by the variable gain amplifier (126) is a constant value

as long as the classification unit (134) determines a level of the audio signals above a given threshold, wherein said constant value corresponds to the stored optimum value.

25. A system for providing hearing assistance to a user (101), comprising a microphone arrangement (26) for capturing audio signals, a transmission unit (102) for transmitting the audio signals via a wireless link (107) to a receiver unit (103) to be worn by user, a variable gain amplifier (126) located in the receiver unit (103) for applying a gain to the audio signals, and a hearing instrument (104) which is to be worn at the user's ear (39) and which is connected to the receiver unit (103) or comprises the receiver unit, said hearing instrument comprising means (38) to be worn at or in the user's ear for stimulating the hearing of the user according to the audio signals from the variable gain amplifier (126), a second microphone arrangement (36) for capturing second audio signals, and means for mixing the audio signals from the variable gain amplifier (126) and the second audio signals prior to stimulating the user's hearing with the mixed audio signals via said stimulating means, **characterized by** means (26, 160) for generating test audio signals and transmitting said test audio signals at a pre-defined level from the transmission unit via the wireless link to the receiver unit; means for simultaneously transmitting gain control commands from the transmission unit to the variable gain amplifier (126) in order to selectively change the gain applied by the variable gain amplifier (126) in order to determine an optimum value of the gain; means (130) for storing said optimum value of the gain; and means for transmitting a store command from the transmission unit to the receiver unit in order to store that determined optimum value of the gain in the storing means.
26. The system of claim 25, wherein the microphone arrangement (26) is integrated into the transmission unit (102).

Patentansprüche

1. Verfahren zum Einstellen eines Systems zur Gehörunterstützung für einen Nutzer (101), welches eine Mikrofonanordnung (26) zum Gewinnen von Audiosignalen, eine Sendeeinheit (102) zum Senden der Audiosignale über eine drahtlose Strecke (107) an eine von dem Nutzer getragene Empfängereinheit (103), einen Verstärker (126) mit variabler Verstärkung, der in der Empfängereinheit angeordnet ist, um die Audiosignale mit einer Verstärkung zu beaufschlagen, sowie eine Hörvorrichtung (104) aufweist,

die am Ohr (39) des Nutzers getragen wird und mit der Empfängereinheit (103) verbunden ist oder diese umfasst, wobei die Hörvorrichtung am oder in einem Ohr (39) des Nutzers getragene Mittel (38) zum Stimulieren des Gehörs des Nutzers gemäß den Audiosignalen von dem Verstärker (126) mit variabler Verstärkung, eine zweite Mikrofonanordnung (36) zum Gewinnen von zweiten Audiosignalen sowie Mittel zum Mischen der Audiosignale von dem Verstärker (126) mit variabler Verstärkung und der zweiten Audiosignale vor dem Stimulieren des Gehörs des Nutzers mit den gemischten Audiosignalen mittels der Stimulationsmittel aufweist, **dadurch gekennzeichnet, dass**

- (a) Testaudiosignale erzeugt und bei einem vorbestimmten Pegel von der Sendeeinheit über die drahtlose Strecke an die Empfängereinheit gesendet werden und das Gehör des Nutzers mit den Testaudiosignalen mittels der Stimulationsmittel stimuliert wird;
- (b) gleichzeitig Verstärkungssteuerbefehle von der Sendeeinheit an den Verstärker mit variabler Verstärkung gesendet werden, um die von dem Verstärker (126) mit variabler Verstärkung angewandte Verstärkung selektiv zu verändern;
- (c) die Schritte (a) und (b) wiederholt werden, bis ein optimaler Verstärkungswert von dem Verstärker (126) mit variabler Verstärkung angewandt wird; und
- (d) ein Speicherbefehl von der Sendeeinheit zu der Empfängereinheit gesendet wird, um diesen bestimmten optimalen Verstärkungswert zu speichern.

2. Verfahren gemäß Anspruch 1, wobei im Schritt (a) die Testsignale **dadurch** erzeugt werden, dass Audiosignale aus einem Audiosignalspeicher (160) abgerufen werden.
3. Verfahren gemäß Anspruch 2, wobei der Audiosignalspeicher (160) in die Sendeeinheit (102) integriert ist.
4. Verfahren gemäß Anspruch 1, wobei im Schritt (a) die Testaudiosignale mittels eines Audiosignalsynthesizers erzeugt werden.
5. Verfahren gemäß Anspruch 4, wobei der Audiosignalsynthesizer in die Sendeeinheit (102) integriert ist.
6. Verfahren gemäß Anspruch 1, wobei im Schritt (a) die Testaudiosignale **dadurch** erzeugt werden, dass Testschall erzeugt wird und der Testschall als die Testaudiosignale mittels der Mikrofonanordnung (26) gewonnen wird.

7. Verfahren gemäß Anspruch 6, wobei es sich bei dem Testschall um die Stimme einer die Sendeeinheit (102) benutzenden Person (100) handelt.
8. Verfahren gemäß einem der Ansprüche 6 und 7, wobei ferner der Testschall mittels der zweiten Mikrofonanordnung (36) als die zweiten Audiosignale gewonnen wird, die Audiosignale von dem Verstärker (126) mit variabler Verstärkung und die zweiten Audiosignale gemäß der momentan eingestellten Verstärkung gemischt werden und das Gehör des Nutzers mit den gemischten Audiosignalen mittels der Stimulationsmittel der Hörvorrichtung stimuliert wird. 5 10
9. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei im Schritt (d) der bestimmte optimal Verstärkungswert in einem Speicher (130) gespeichert wird, der in die Empfängereinheit (103) oder die Hörvorrichtung (104) integriert ist. 15 20
10. Verfahren gemäß Anspruch 9, wobei vor dem Schritt (a) die Empfängereinheit (103) identifiziert wird, indem in der Empfängereinheit gespeicherte Identifikationsinformation mittels der Sendeeinheit (102) über eine induktive Strecke (54) gelesen wird. 25
11. Verfahren gemäß Anspruch 10, wobei die Empfängereinheit (103) von der Sendeeinheit (102) spezifisch adressiert wird, indem ein gemäß der von der Sendeeinheit gelesenen Identifikationsinformation kodiertes Signal gesendet wird. 30
12. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei im Schritt (a) das Testsignal bei einem maximalen Pegel der Audiosignale der Sendeeinheit (102) gesendet wird. 35
13. Verfahren gemäß Anspruch 1, wobei die Empfängereinheit (103) mit der Hörvorrichtung (104) verbunden ist und der Verstärker (126) mit variabler Verstärkung verstärkungsgesteuert und/oder ausgangsimpedanzgesteuert ist. 40
14. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei eine Datenstrecke zum Senden der Verstärkungssteuerbefehle und dem Speicherbefehl sowie die Audiosignalstrecke (106) mittels eines gemeinsamen Sendekanals realisiert werden. 45
15. Verfahren gemäß Anspruch 14, wobei der untere Bereich der Bandbreite des Sendekanals von der Audiosignalstrecke (106) genutzt wird und der obere Bereich der Bandbreite des Kanals von der Datenstrecke genutzt wird. 50 55
16. Verfahren gemäß Anspruch 1, wobei der Ausgang der Empfängereinheit (103) mit der zweiten Mikrofonanordnung (36) parallel geschaltet ist.
17. Verfahren gemäß Anspruch 1, wobei die Audiosignale der Hörvorrichtung (104) über einen von der zweiten Mikrofonanordnung (36) getrennten Audioeingang von der Empfängereinheit (103) zugeführt werden.
18. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei es sich bei der Audiosignalstrecke um eine frequenzmodulierte Funkstrecke handelt.
19. Verfahren gemäß Anspruch 1, wobei es sich bei der Hörvorrichtung (104) um ein Hörgerät mit einem elektroakustischen Ausgangswandler (38) als die Stimulationsmittel handelt.
20. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei die Audiosignale in der Sendeeinheit (102) in einer Verstärkungsmodelleinheit (112) einer Verarbeitung mit automatischer Verstärkungssteuerung unterzogen werden, bevor sie zu der Empfängereinheit gesendet werden.
21. Verfahren gemäß einem der vorhergehenden Ansprüche, wobei es sich bei der von dem Verstärker (126) mit variabler Verstärkung abgewandten Verstärkung um einen konstanten Wert handelt, welcher dem gespeicherten optimalen Verstärkungswert entspricht.
22. Verfahren gemäß einem der Ansprüche 1 bis 21, wobei ferner
 - (a) mittels der Mikrofonanordnung (26) Audiosignale gewonnen werden und die Audiosignale mittels der Sendeeinheit (102) über die drahtlose Audiodatenstrecke (107) an die Empfängereinheit (103) gesendet werden;
 - (b) die Audiosignale vor dem Senden mittels einer Klassifikationseinheit (134) analysiert werden, um eine momentane Hörumgebungskategorie aus einer Mehrzahl von Hörumgebungskategorien zu bestimmen;
 - (c) die Audiosignale mittels des Verstärkers (126) mit variabler Verstärkung mit einer Verstärkung beaufschlagt werden, die gemäß der in Schritt (b) bestimmten momentanen Hörumgebungskategorie ausgewählt ist;
 - (d) das Gehör des Nutzers mittels der Stimulationsmittel (38) gemäß den Audiosignalen von dem Verstärker (126) mit variabler Verstärkung stimuliert wird;
 wobei der gespeicherte optimale Verstärkungswert verwendet wird, um den Verstärker (126) mit variabler Verstärkung zu kalibrieren.
23. Verfahren gemäß Anspruch 22, wobei es sich bei der für mindestens eine der Umgebungen ange-

wandten Verstärkung um den gespeicherten optimalen Verstärkungswert handelt.

24. Verfahren gemäß Anspruch 23, wobei es sich bei der von dem Verstärker (126) mit variabler Verstärkung angewendeten Verstärkung um einen konstanten Wert handelt, solange die Klassifikationseinheit (134) einen Pegel der Audiosignale oberhalb eines vorgegebenen Schwellwerts erfasst, wobei der konstante Wert dem gespeicherten optimalen Wert entspricht.
25. System zur Hörunterstützung für einen Nutzer (101), mit einer Mikrofonanordnung (26) zum Gewinnen von Audiosignalen, einer Sendeeinheit (102) zum Senden der Audiosignale an eine von dem Nutzer zu tragenden Empfängereinheit (103) über eine drahtlose Strecke (107), einem in der Empfängereinheit (103) angeordneten Verstärker (126) mit variabler Verstärkung zum Beaufschlagen der Audiosignale mit einer Verstärkung, sowie einer am Ohr (39) des Nutzers zu tragenden Hörvorrichtung (104), die mit der Empfängereinheit (103) verbunden ist oder diese Empfängereinheit umfasst, wobei die Hörvorrichtung am oder im Ohr des Nutzers zu tragende Mittel (38) zum Stimulieren des Gehörs des Nutzers gemäß den Audiosignalen von dem Verstärker (126) mit variabler Verstärkung, eine zweite Mikrofonanordnung (36) zum Gewinnen von zweiten Audiosignalen sowie Mittel zum Mischen der Audiosignale von dem Verstärker (126) mit variabler Verstärkung und der zweiten Audiosignale vor dem Stimulieren des Gehörs des Nutzers mit den gemischten Audiosignalen mittels der Stimulationsmittel aufweist, **gekennzeichnet durch** Mittel (26, 160) zum Erzeugen von Testaudiosignalen und Senden der Testaudiosignale bei einem vorbestimmten Pegel von der Sendeeinheit zu der Empfängereinheit über die drahtlose Strecke; Mittel zum gleichzeitigen Senden von Verstärkungssteuerbefehlen von der Sendeeinheit zu dem Verstärker (126) mit variabler Verstärkung, um die von dem Verstärker (126) mit variabler Verstärkung angewandte Verstärkung selektiv zu verändern, um einen optimalen Verstärkungswert zu bestimmen; Mittel (130) zum Speichern des optimalen Verstärkungswerts; und Mittel zum Senden eines Speicherbefehls von der Sendeeinheit zu der Empfängereinheit, um diesen bestimmten optimalen Verstärkungswert in den Speichermitteln zu speichern.
26. System gemäß Anspruch 25, wobei die Mikrofonanordnung (26) in die Sendeeinheit (102) integriert ist.

Revendications

- Procédé pour ajuster un système destiné à fournir une aide auditive à un utilisateur (101), le système comprenant un agencement de microphone (26) pour capturer des signaux audio, une unité de transmission (102) pour transmettre les signaux audio au moyen d'une liaison sans fil (107) à une unité de réception (103) portée par l'utilisateur, un amplificateur (126) à gain variable placé dans l'unité de réception pour appliquer un gain aux signaux audio, et un instrument auditif (104) qui est porté à l'oreille (39) de l'utilisateur et qui est relié à l'unité de réception (103) ou qui comprend l'unité de réception, ledit instrument auditif comprenant un moyen (38) porté à ou dans l'oreille (39) de l'utilisateur pour stimuler l'audition de l'utilisateur suivant les signaux audio provenant de l'amplificateur (126) à gain variable, un deuxième agencement de microphone (36) pour capturer des deuxième signaux audio, et un moyen pour mélanger les signaux audio provenant de l'amplificateur (126) à gain variable et les deuxième signaux audio avant de stimuler l'audition de l'utilisateur avec les signaux audio mélangés par ledit moyen de stimulation, **caractérisé par le fait** :
 - de générer des signaux audio de test, de transmettre lesdits signaux audio de test de l'unité de transmission à un niveau prédéfini au moyen de la liaison sans fil à l'unité de réception et de stimuler l'audition de l'utilisateur avec lesdits signaux audio de test par l'intermédiaire dudit moyen de stimulation ;
 - de transmettre simultanément des ordres de commande de gain de l'unité de transmission à l'amplificateur à gain variable afin de changer au choix le gain appliqué par l'amplificateur (126) à gain variable ;
 - de répéter les étapes (a) et (b) jusqu'à ce qu'une valeur optimale du gain soit appliquée par l'amplificateur (126) à gain variable ; et
 - de transmettre un ordre de stockage de l'unité de transmission à l'unité de réception afin de stocker cette valeur optimale déterminée du gain.
- Procédé de la revendication 1, dans lequel, à l'étape (a), lesdits signaux audio de test sont générés en récupérant des signaux audio d'une mémoire (160) de signaux audio.
- Procédé de la revendication 2, dans lequel ladite mémoire (160) de signaux audio est intégrée dans l'unité de transmission (102).
- Procédé de la revendication 1, dans lequel, à l'étape (a), lesdits signaux audio de test sont générés par un synthétiseur de signaux audio.

5. Procédé de la revendication 4, dans lequel ledit synthétiseur de signaux audio est intégré dans l'unité de transmission (102).
6. Procédé de la revendication 1, dans lequel, à l'étape (a), lesdits signaux audio de test sont générés en générant un son de test et en capturant ledit son de test comme étant lesdits signaux audio de test par l'agencement de microphone (26). 5
7. Procédé de la revendication 6, dans lequel ledit son de test est la voix d'une personne (100) utilisant l'unité de transmission (102). 10
8. Procédé de l'une des revendications 6 et 7, comprenant en outre le fait : de capturer ledit son de test comme étant lesdits deuxièmes signaux audio par le deuxième agencement de microphone (36), de mélanger les signaux audio provenant de l'amplificateur (126) à gain variable et les deuxièmes signaux audio selon le gain présentement réglé et de stimuler l'audition de l'utilisateur avec lesdits signaux audio mélangés par l'intermédiaire dudit moyen de stimulation de l'instrument auditif. 15 20
9. Procédé de l'une des revendications précédentes, dans lequel, à l'étape (d), la valeur optimale déterminée du gain est stockée dans une mémoire (130) qui est intégrée dans l'unité de réception (103) ou dans l'instrument auditif (104). 25 30
10. Procédé de la revendication 9, dans lequel, avant l'étape (a), l'unité de réception (103) est identifiée en lisant des informations d'identification stockées dans l'unité de réception par l'unité de transmission (102) via une liaison inductrice (54). 35
11. Procédé de la revendication 10, dans lequel l'unité de réception (103) est spécifiquement adressée par l'unité de transmission (102) en transmettant un signal codé selon les informations d'identification lues par l'unité de transmission. 40
12. Procédé de l'une des revendications précédentes, dans lequel, à l'étape (a), le signal de test est transmis à un niveau maximal des signaux audio de l'unité de transmission (102). 45
13. Procédé de la revendication 1, dans lequel l'unité de réception (103) est reliée à l'instrument auditif (104) et l'amplificateur (126) à gain variable est commandé en gain et/ou en impédance de sortie. 50
14. Procédé de l'une des revendications précédentes, dans lequel une liaison de données pour transmettre les ordres de commande de gain et l'ordre de stockage, et la liaison (106) de signaux audio sont réalisées par une voie de transmission commune. 55
15. Procédé de la revendication 14, dans lequel la partie inférieure de la bande passante de la voie de transmission est utilisée par la liaison (106) de signaux audio et la partie supérieure de la bande passante de la voie est utilisée par la liaison de données.
16. Procédé de la revendication 1, dans lequel la sortie de l'unité de réception (103) est reliée en parallèle au deuxième agencement de microphone (36).
17. Procédé de la revendication 1, dans lequel les signaux audio provenant de l'unité de réception (103) sont fournis à l'instrument auditif (104) par l'intermédiaire d'une entrée audio séparée du deuxième agencement de microphone (36).
18. Procédé de l'une des revendications précédentes, dans lequel la liaison de signaux audio est une liaison radio FM.
19. Procédé de la revendication 1, dans lequel l'instrument auditif (104) est une aide auditive ayant un transducteur (38) de sortie électroacoustique qui fait office du moyen de stimulation.
20. Procédé de l'une des revendications précédentes, dans lequel les signaux audio dans l'unité de transmission (102) subissent un traitement de commande de gain automatique dans une unité (112) de modèle de gain avant d'être transmis à l'unité de réception.
21. Procédé de l'une des revendications précédentes, dans lequel le gain appliqué par l'amplificateur (126) à gain variable est une valeur constante, ladite valeur constante correspondant à la valeur optimale stockée du gain.
22. Procédé de l'une des revendications 1 à 21, comprenant en outre le fait :
 - (a) de capturer des signaux audio par l'agencement de microphone (26) et de transmettre les signaux audio par l'unité de transmission (102) au moyen de la liaison (107) audio sans fil à l'unité de réception (103) ;
 - (b) d'analyser les signaux audio avant d'être transmis par une unité de classification (134) afin de déterminer une catégorie de scènes auditives actuelle à partir d'une pluralité de catégories de scènes auditives ;
 - (c) d'appliquer un gain par l'amplificateur (126) à gain variable aux signaux audio, lequel gain est sélectionné selon la catégorie de scènes auditives actuelle déterminée à l'étape (b) ;
 - (d) de stimuler l'audition de l'utilisateur par le moyen de stimulation (38) suivant les signaux audio provenant de l'amplificateur (126) à gain variable ;

dans lequel la valeur optimale stockée du gain est utilisée pour étalonner l'amplificateur (126) à gain variable.

23. Procédé de la revendication 22, dans lequel le gain appliqué pour au moins l'une des scènes auditives est la valeur optimale stockée du gain. 5

24. Procédé de la revendication 23, dans lequel le gain appliqué par l'amplificateur (126) à gain variable est une valeur constante tant que l'unité de classification (134) détermine un niveau des signaux audio au-dessus d'un seuil donné, où ladite valeur constante correspond à la valeur optimale stockée. 10
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25. Système pour fournir une aide auditive à un utilisateur (101), comprenant un agencement de microphone (26) pour capturer des signaux audio, une unité de transmission (102) pour transmettre les signaux audio par l'intermédiaire d'une liaison sans fil (107) à une unité de réception (103) à porter par l'utilisateur, un amplificateur (126) à gain variable placé dans l'unité de réception (103) pour appliquer un gain aux signaux audio, et un instrument auditif (104) qui doit être porté à l'oreille (39) de l'utilisateur et qui est relié à l'unité de réception (103) ou qui comprend l'unité de réception, ledit instrument auditif comprenant un moyen (38) à porter à ou dans l'oreille de l'utilisateur pour stimuler l'audition de l'utilisateur selon les signaux audio provenant de l'amplificateur (126) à gain variable, un deuxième agencement de microphone (36) pour capturer des deuxièmes signaux audio, et un moyen pour mélanger les signaux audio provenant de l'amplificateur (126) à gain variable et les deuxièmes signaux audio avant de stimuler l'audition de l'utilisateur avec les signaux audio mélangés par l'intermédiaire dudit moyen de stimulation, **caractérisé par** 20
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 - un moyen (26, 160) pour générer des signaux audio de test et pour transmettre lesdits signaux audio de test à un niveau prédéfini de l'unité de transmission via la liaison sans fil à l'unité de réception ;
 - un moyen pour transmettre simultanément des ordres de commande de gain de l'unité de transmission à l'amplificateur (126) à gain variable afin de changer au choix le gain appliqué par l'amplificateur (126) à gain variable afin de déterminer une valeur optimale du gain ;
 - un moyen (130) pour stocker ladite valeur optimale du gain ; et
 - un moyen pour transmettre un ordre de stockage de l'unité de transmission à l'unité de réception afin de stocker cette valeur optimale déterminée du gain dans le moyen de stockage.

26. Système de la revendication 25, dans lequel l'agencement de microphone (26) est intégré dans l'unité de transmission (102).

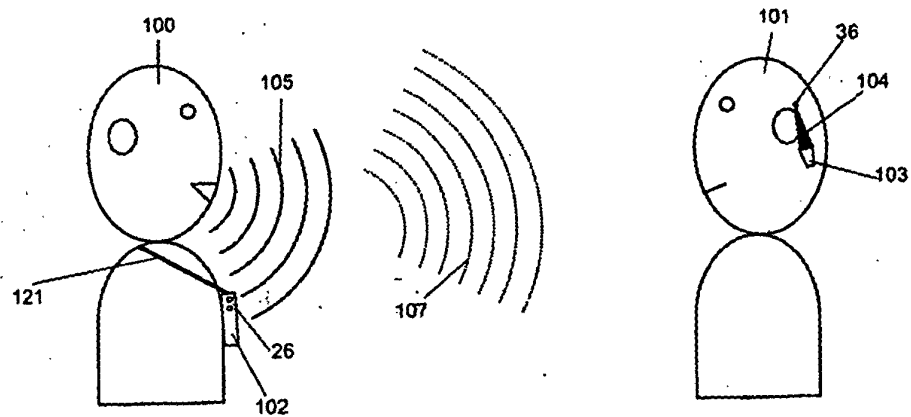


Fig. 1

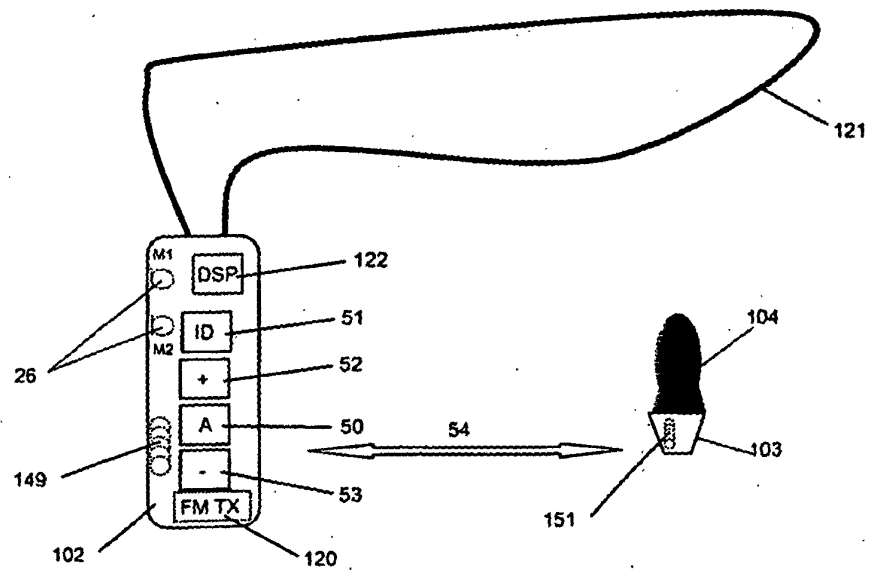


Fig. 2

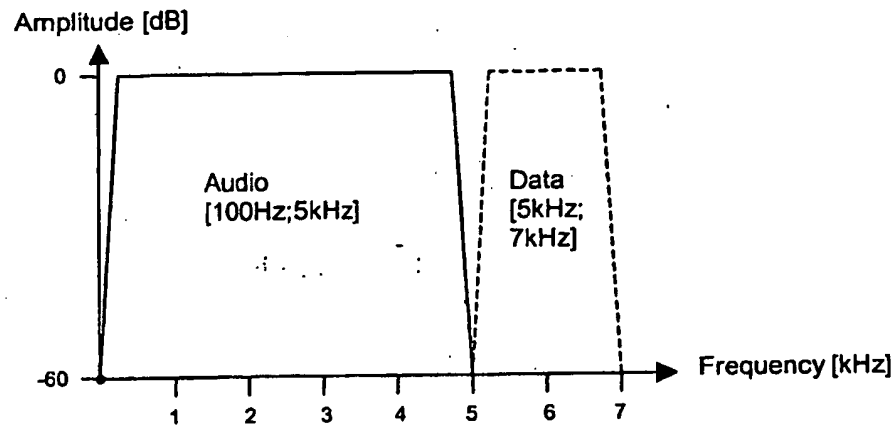


Fig. 3

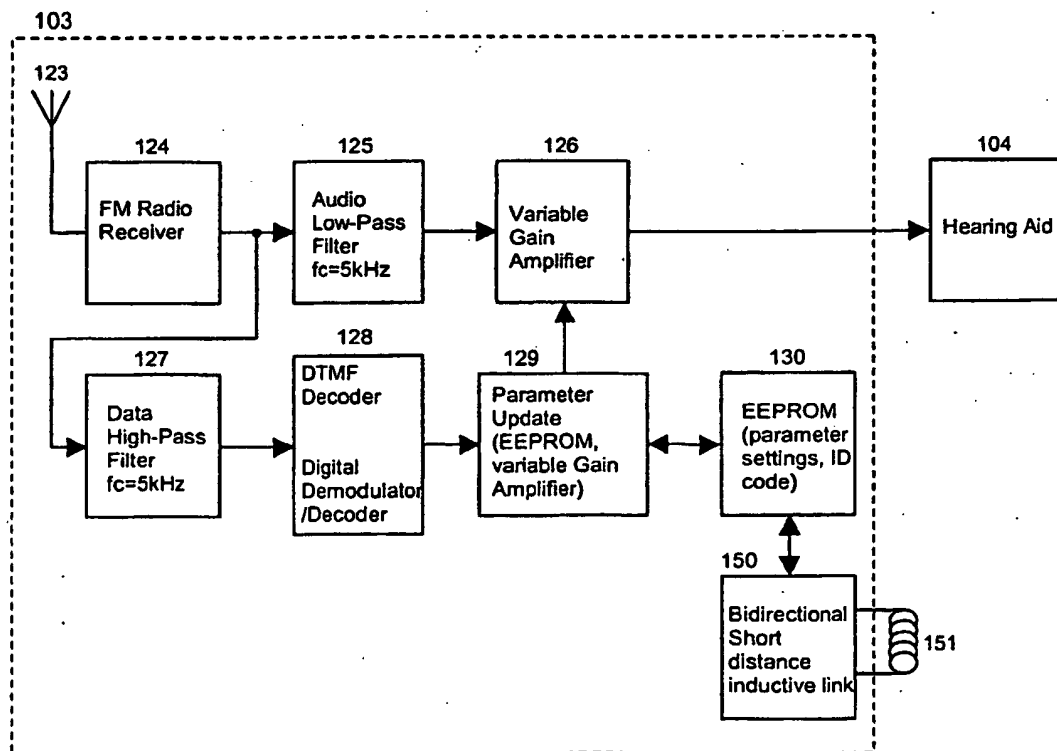


Fig. 4

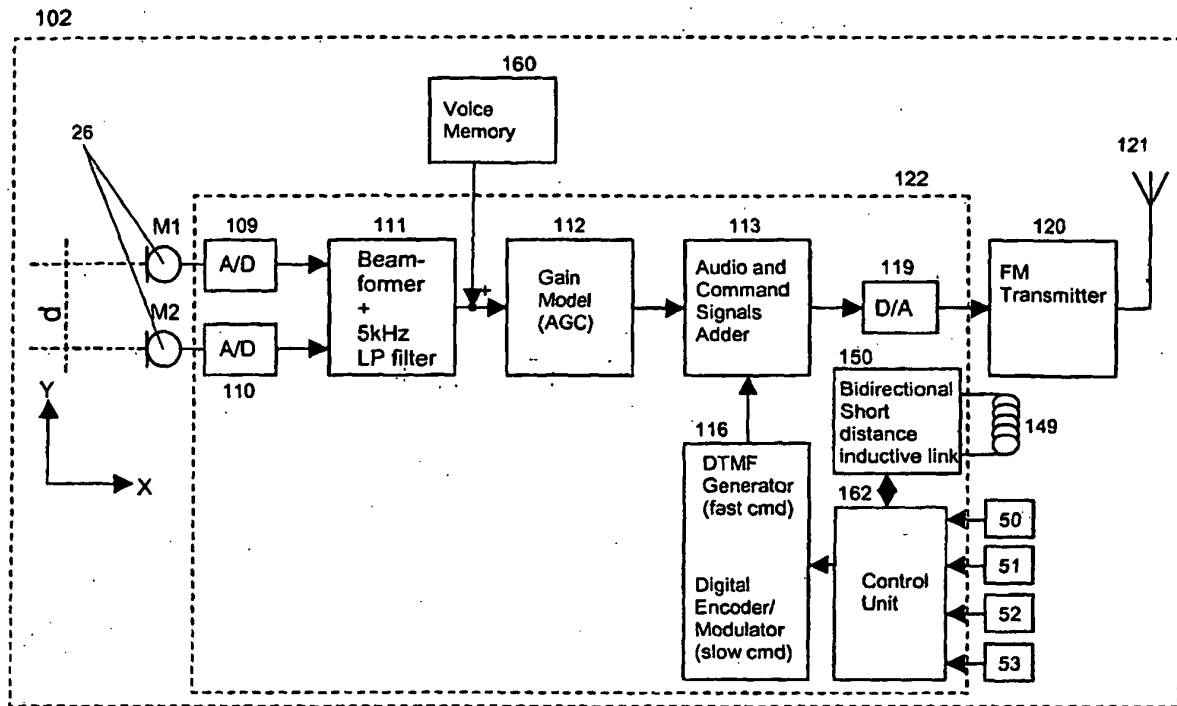


Fig. 5

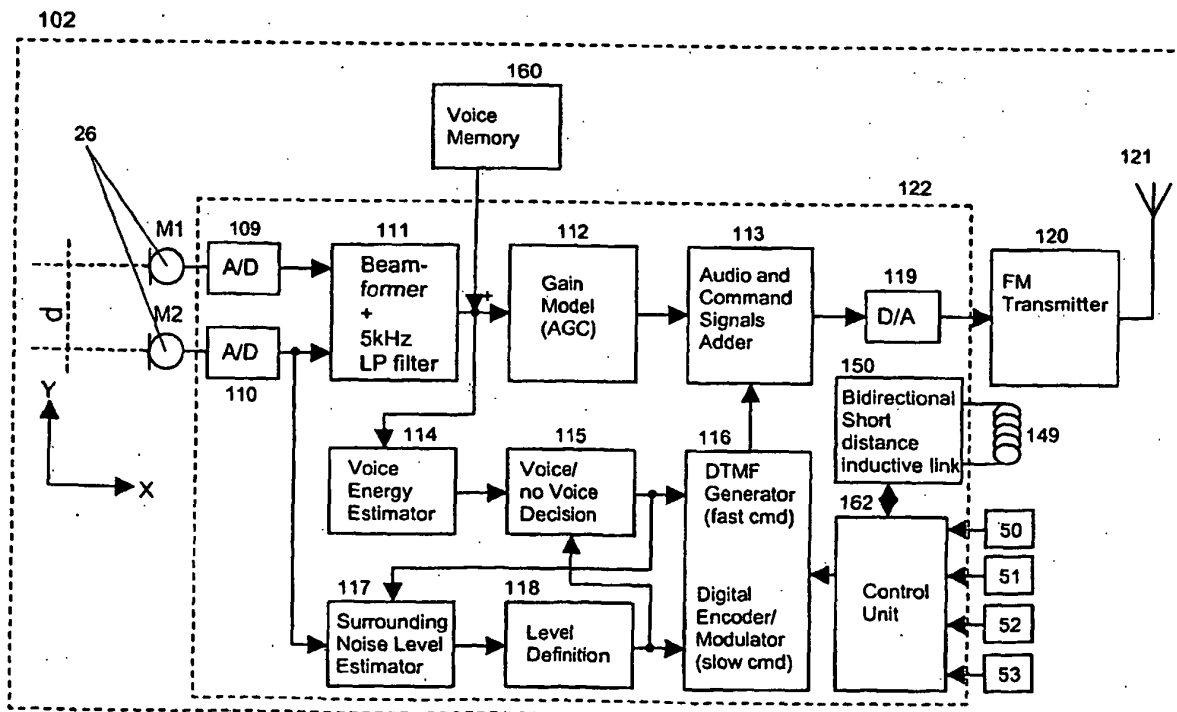


Fig. 6

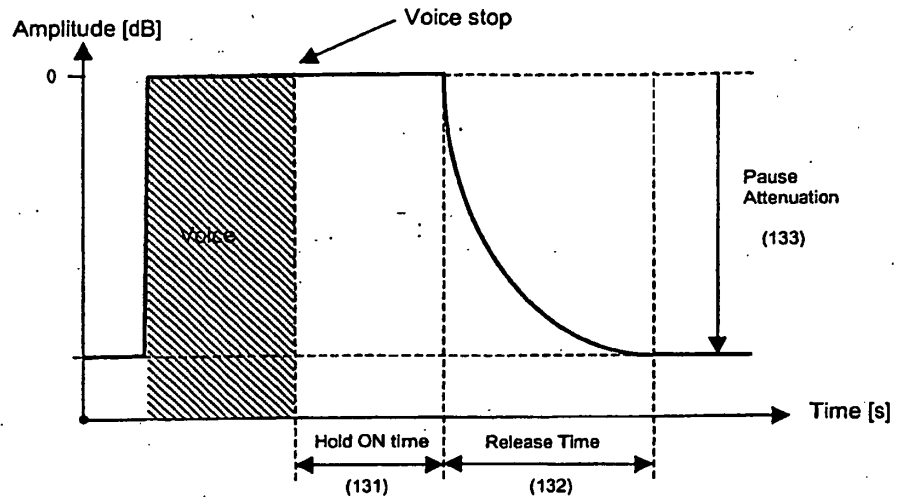


Fig. 7

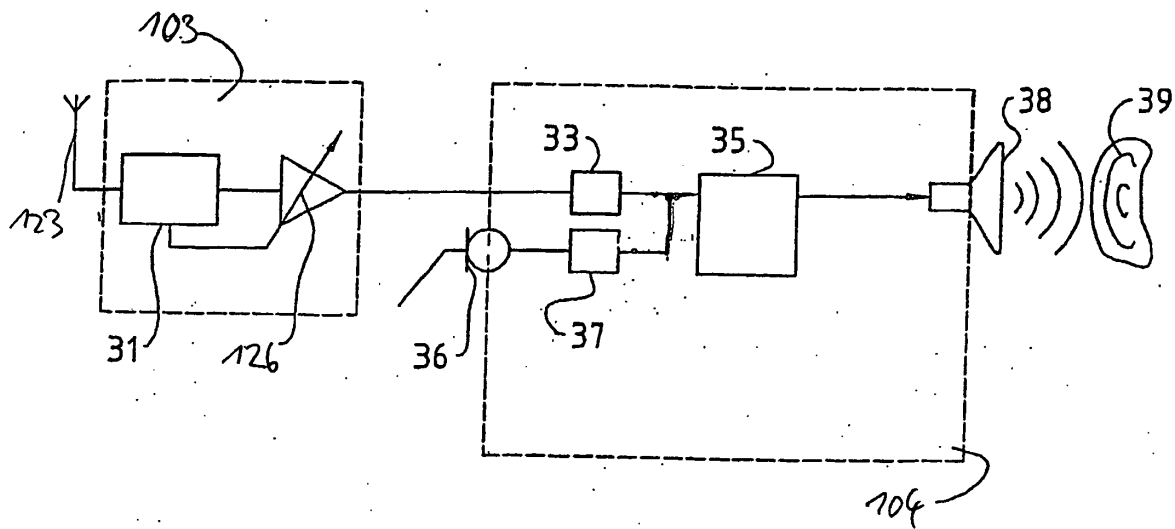


Fig. 8

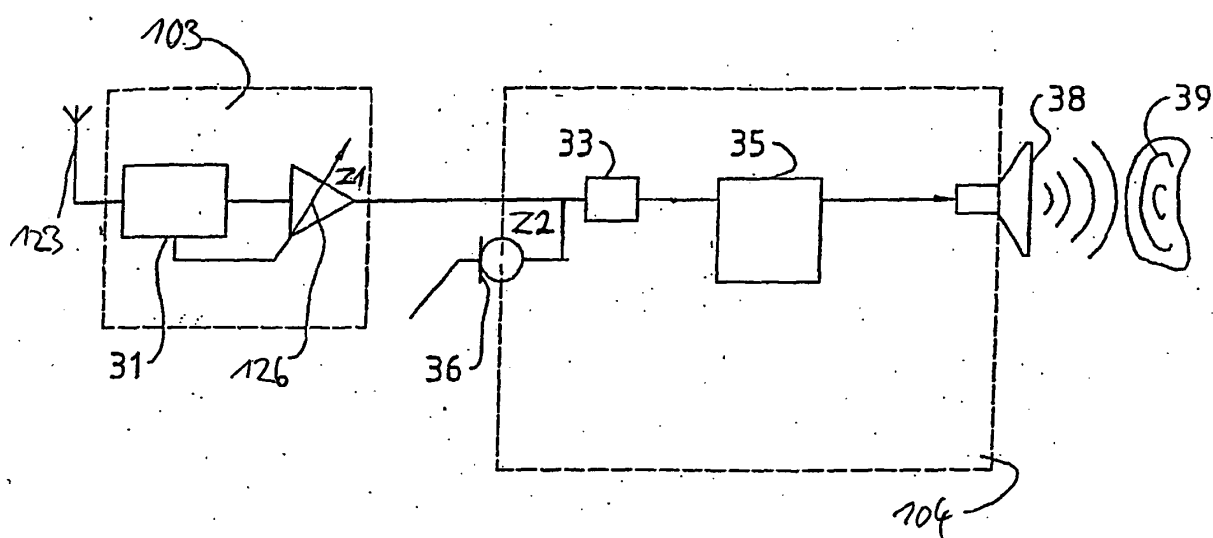


Fig. 9

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

- WO 0223948 A1 [0006]
- EP 1638367 A2 [0007]
- WO 9721325 A1 [0008]
- WO 0230153 A1 [0009]
- US 20020037087 A [0010]
- US 20020090098 A [0010]
- WO 02032208 A [0010]
- US 20020150264 A [0010]
- DE 10345173 B3 [0011]
- US 20030235319 A [0012]
- EP 1443803 A2 [0013]
- DE 102004025691 B3 [0014]
- EP 0671818 B1 [0015]
- EP 1619926 A1 [0015]