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(54) Title: ADAPTABILITY TO MULTIPLE HANDS-FREE COMMUNICATION ACCESSORIES

(57) Abstract: To optimize the performance of a device, such as a mobile telephone, that may be coupled to any of a variety of hands-free adapters, each characterized by a particular quality of performance, the device receives self-identifying information from the particular hands-free adapter to which it is coupled. The device then determines one or more settings as a function of the self-identifying information, and adjusts one or more components in accordance with the one or more settings. For example, such adjustments may include microphone gain settings, loudspeaker gain settings, and adjustments to acoustic echo canceler equipment.

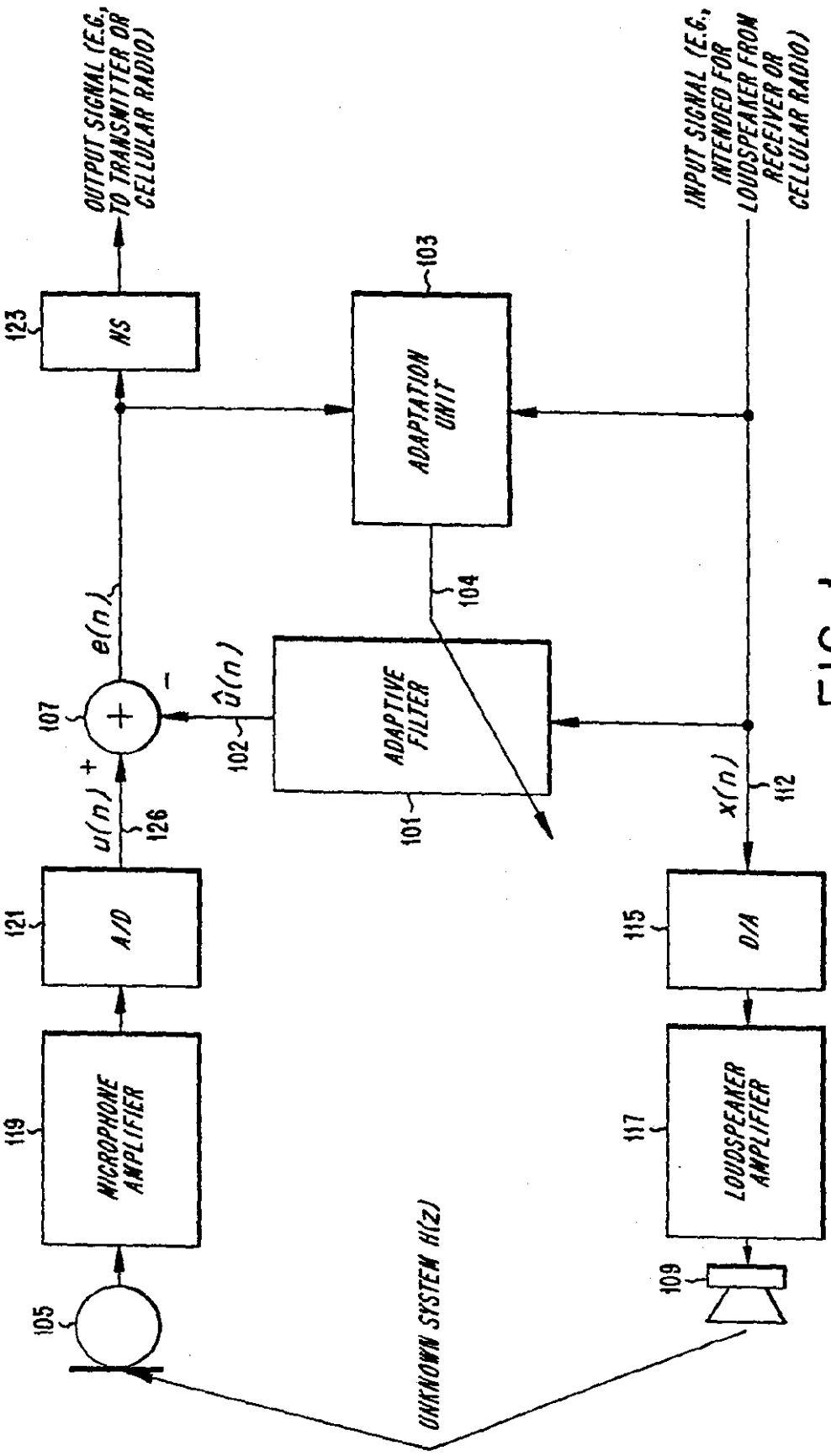



FIG. 1
(PRIOR ART)

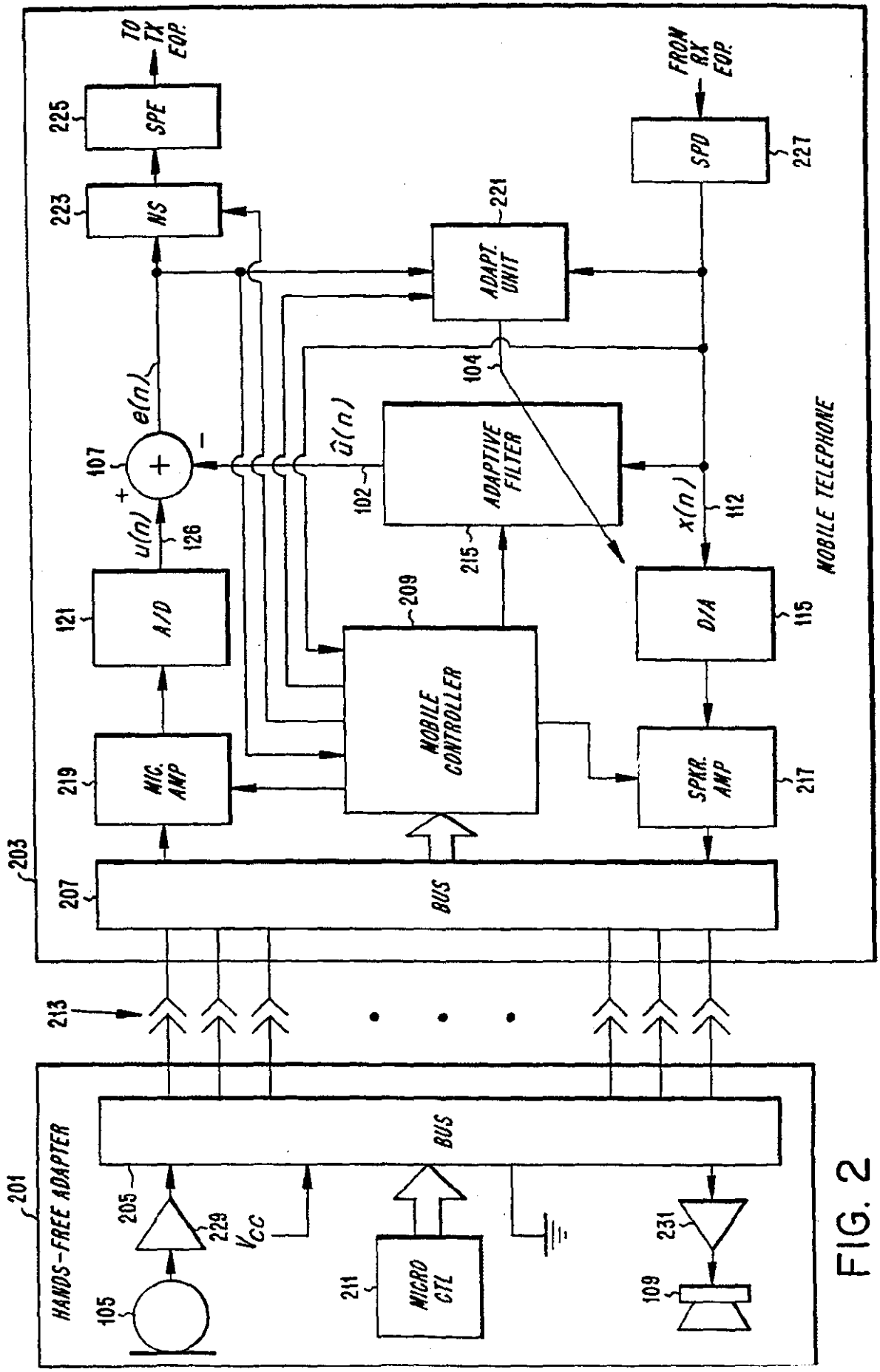

HANDS-FREE ADAPTER
201
MOBILE TELEPHONE
203
BUS
205
BUS
207
213
MICRO CTL
211
MOBILE CONTROLLER
209
ADAPTIVE FILTER
102
ADAPT. UNIT
221
MIC. AMP
219
SPKR. AMP
217
A/D
121
D/A
115
SPE
225
NS
223
SPD
227
TO TX EQP.
FROM RX EQP.
105
229
231
109
107
e(n)
u(n)
126
x(n)
112
104
215

FIG. 2

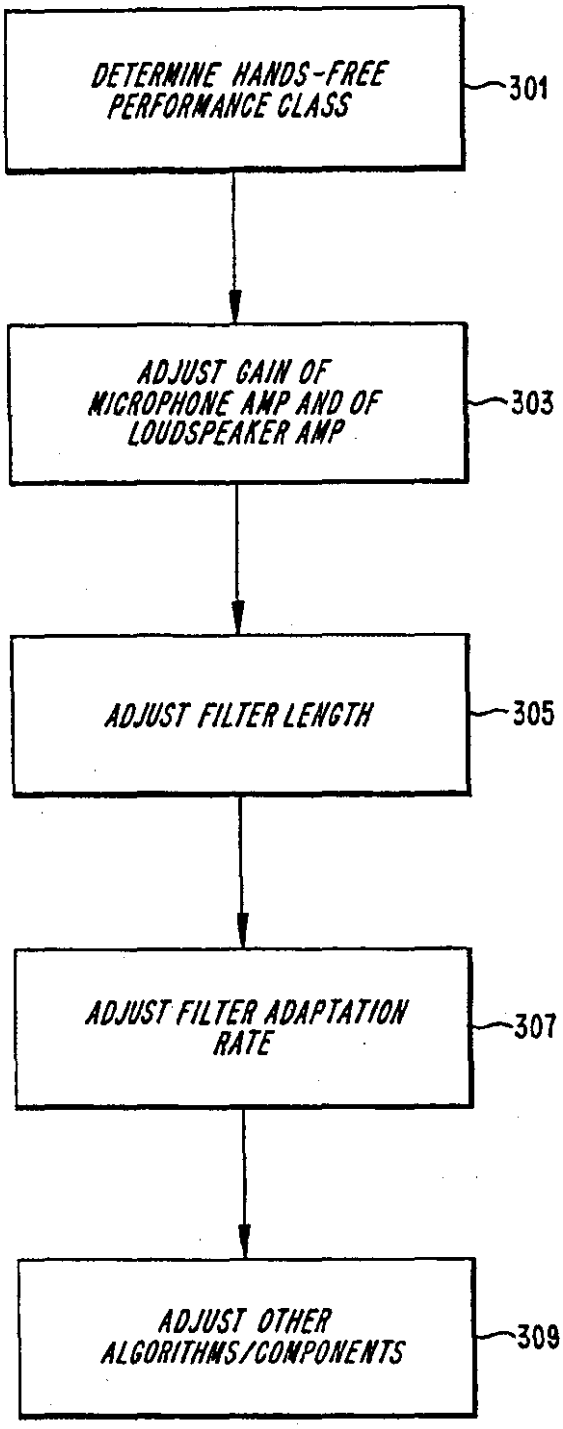

DETERMINE HANDS-FREE PERFORMANCE CLASS
301
ADJUST GAIN OF MICROPHONE AMP AND OF LOUDSPEAKER AMP
303
ADJUST FILTER LENGTH
305
ADJUST FILTER ADAPTATION RATE
307
ADJUST OTHER ALGORITHMS/COMPONENTS
309

FIG. 3

# ADAPTABILITY TO MULTIPLE HANDS-FREE COMMUNICATION ACCESSORIES

## BACKGROUND

The present invention relates telecommunications and other audio devices, and more particularly to the use of hands-free accessories in conjunction with telecommunications and other audio devices.

In the field of telecommunications, such as with speaker phones and in cellular telephony, it is often desirable to allow a user to operate communication equipment without requiring the continued occupation of one or more of the user's hands. This can be an important factor in environments, such as automobiles, where a driver's preoccupation with holding telephone equipment may jeopardize not only his or her safety, but also the safety of others who share the road. Freedom to use one's hands for something other than holding a microphone is useful in other applications as well, such as with internet communication by means of a personal computer, speech recognition by a computer, or with audio-visual presentation systems.

To accommodate these important needs, so-called "hands-free" equipment has been developed, in which microphones and loudspeakers are mounted or otherwise placed within the hands-free environment, thereby obviating the need to hold them. For example, in an automobile application, a cellular telephone's microphone might be mounted on the sun visor, while the loudspeaker may be a dash-mounted unit, or may be one that is associated with the car's stereo equipment. With components mounted in this fashion, a cellular telephone user may carry on a conversation without having to hold the cellular unit or its handset. Similarly, personal computers often have microphones and loudspeakers mounted, for example, in a monitor in relatively close proximity to each other.

One problem with a hands-free arrangement is that the microphone tends to pick up sound from the nearby loudspeaker, in addition to the voice of the user of the hands-free equipment (the so-called "near-end user"). This is also a problem in some non-hands-free devices, such as handheld mobile telephones, which are becoming

smaller and smaller. (Because of the small size, a mobile telephone's microphone cannot entirely be shielded from the sound emitted by its loudspeaker.) This sensing by the microphone of sound generated by the loudspeaker can cause problems in many types of applications. For example, in communications equipment, delays introduced by the communications system as a whole can cause the sound from the loudspeaker to be heard by the individual on the other end of the call (the so-called "far-end") as an echo of his or her own voice. Such an echo degrades audio quality and its mitigation is desirable if not an absolute necessity because the presence of a delayed echo seriously impedes the far-end user's ability to carry on a normal conversation. A similar problem can exist, for example, in automated systems that synthesize speech through a loudspeaker, and include voice recognition components for recognizing and responding to spoken commands or other words sensed by the microphone. In such applications, the presence of an echo of synthesized speech in the microphone signal can severely degrade the performance of the speech recognition components. Solutions for ameliorating such echoes include utilizing an adaptive echo cancellation filter, an echo attenuator or both in some kind of combined operation.

As a representative example of hands-free equipment in general, an exemplary "hands-free" mobile telephone, having a conventional echo canceler in the form of an adaptive filter arrangement, is depicted in FIG. 1. A hands-free communications environment may be, for example, an automotive interior in which the mobile telephone is installed. Such an environment can cause effects on an acoustic signal propagating therein, which effects are typically unknown. Henceforth, this type of environment will be referred to throughout this specification as an unknown system H(z). The hands-free mobile telephone includes a loudspeaker 109 for supplying far-end acoustic audio signals to the near-end user. The hands-free mobile telephone also includes a microphone 105 for converting near-end acoustic signals (usually generated by the near-end user) into an electrical or other signal for transmission to the far-end user.

In operation, a digital input audio signal 112 representing, say, a far-end user's voice is provided to a digital-to-analog (D/A) converter 115. The resultant analog signal is suitably amplified by a loudspeaker amplifier 117 before being supplied to the loudspeaker 109. In the reverse direction, the microphone 105 receives acoustic

energy and converts this into an analog signal. This signal is suitably amplified by microphone amplifier 119, and then converted into a series of digital samples by analog-to-digital converter 121. (It is noted that in the exemplary embodiment, most of the processing is performed on digital signals, hence the need for the D/A and A/D converters 115, 121. In other embodiments and other applications, however, all of the processing depicted herein could be performed on analog signals. In such embodiments, the D/A and A/D converters 115, 121 could be dispensed with.) The resultant signal 126 is then further processed as necessary for eventual transmission to the far-end user.

In this arrangement, the microphone 105 is intended for detecting a user's voice, but may also have the undesired effect of detecting audio signals emanating from the loudspeaker 109. It is this undesired action that introduces the echo signal into the system.

Circuitry for reducing, if not eliminating, the echo includes an adaptive filter 101, such as an adaptive Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter, an adaptation unit 103, such as a least mean square (LMS) cross correlator, and a subtractor 107. If one assumes that the only sound received by the microphone 105 is that originating from the loudspeaker 109 (of course, in practice the microphone will also receive sounds from the intended user as well as background noise), then a digitized echo signal 126 (also designated "u") is generated. In operation, the adaptive filter 101 generates an echo estimate signal 102, which is commonly referred to as a û signal. The echo estimate signal 102 may be formed as the convolution of the far-end signal 112, and a sequence of $m$ filter weighting coefficients ($h_i$) of the filter 101 (See Equation 1).

$$\hat{u}(n)=\sum_{i\in}^{1-0} h_i \cdot x(n-i) \qquad (1)$$

where:

x(n) is the input signal,

m is the number of weighting coefficients, and

n is the sample number.

When the weighting coefficients are set correctly, the adaptive filter 101 produces an impulse response that is approximately equal to the response produced by the loudspeaker 109 within the unknown system H(z). The echo estimate signal 102 generated by the adaptive filter 101 is subtracted from the incoming digitized echo signal 126, to produce an echo filtered microphone signal, also referred to as an error signal $e(n)$ because its energy is entirely attributable to the error in the echo estimate signal 102 when the only sound reaching the microphone 105 is attributable to the sound emanating from the loudspeaker 109:

$$e(n)=u(n)-\hat{u}(n) \tag{2}$$

Ideally, any echo response from the unknown system H(z), introduced by the loudspeaker 109, is removed from the digitized echo signal 126 by the subtraction of the echo estimate signal 102. Consequently, when the digitized echo signal 126 also includes signal components attributable to the intended near-end user, the resulting difference signal should ideally include only these near-end user components. Typically, the number of weighting coefficients (henceforth referred to as "coefficients") required for effectively canceling an echo will depend on the application. For handheld phones, fewer than one hundred coefficients may be adequate. For a hands-free telephone in an automobile, about 200 to 400 coefficients will be required. A large room may require a filter utilizing over 1000 coefficients in order to provide adequate echo cancellation.

It can be seen that the effectiveness of the echo canceler is directly related to how well the adaptive filter 101 is able to replicate the impulse response of the unknown system H(z). This, in turn, is directly related to the set of coefficients, $h_i$, maintained by the filter 101.

It is advantageous to provide a mechanism for dynamically altering the coefficients, $h_i$, to allow the adaptive filter 101 to adapt to changes in the unknown system H(z). In a car having a hands-free cellular arrangement, such changes may

occur when a window or car door is opened or closed. In fact, in this and other environments as well, changes in the unknown system H(z) can also result from any number of changes in the environment, such as movement of a person or object, or a change in the relative position or orientation of the microphone 105 and loudspeaker 109 with respect to one another. A well-known coefficient adaptation scheme is the Least Mean Square (LMS) process, which was first introduced by Widrow and Hoff in 1960, and is frequently used because of its efficiency and robust behavior. As applied to the echo cancellation problem, the LMS process is a stochastic gradient step method which uses a rough (noisy) estimate of the gradient, $g(n) = e(n) \cdot \underline{x}(n-1)$, to make an incremental step toward minimizing the energy (and the mean square error) of the error signal, $e(n)$, where $\underline{x}(n)$ is in vector notation corresponding to an expression $\underline{x}(n)=[x(n)\ x(n-1)\ x(n-2)\ \cdots\ x(n-m+1)]$. The update information produced by the LMS process $e(n) \cdot \underline{x}(n)$ is used to determine the value of a coefficient in a next sample. The expression for calculating a next coefficient value $h_i(n+1)$ is given by:

$$h_i(n+1)=h_i(n)+\mu e(n)\cdot x(n-i), \qquad i=0..m-1 \tag{3}$$

where

x(n) is an $n$th sample of the digitized input signal 112,

$h_i(n)$ is a filter weighting coefficient,

$i$ designates a particular coefficient,

$m$ is the number of coefficients,

$n$ is the sample number, and

$\mu$ is a step or update gain parameter.

The LMS method is very common in real-time applications due to its efficiency and simplicity. It produces information in incremental portions each of which portions may have a positive or a negative value. The information produced by the LMS process can be provided to the adaptive filter 101 to update the filter's coefficients.

Referring back to FIG. 1, the exemplary conventional echo cancellation circuit includes a filter adaptation unit 103 in the form of an LMS cross correlator for providing coefficient update information 104 to the filter 101. In this arrangement, the filter adaptation unit 103 monitors the error signal $e(n)$ that represents the digitized echo signal 126 minus the echo estimate signal 102 generated by the filter 101. The echo estimate signal 102 is generated, as described above, with the use of update information 104 provided to the adaptive filter 101 by the filter adaptation unit 103. The coefficients, $h_i$, of the adaptive filter 101 accumulate the update information 104 as shown by Eq. (3).

Despite the presence of echo cancellation circuitry, such as that described above, the signals generated for further processing (e.g., for transmission to the far-end user in a communications system, or for near-end speech recognition or for controlling the operation of the echo cancellation filter 101) may very often still include echo and/or other noise components. This may occur, for example, because the adaptive filter has not yet converged to a fully adapted state, or even after such convergence whenever the unknown environment H(z) changes, thereby requiring the adaptation process to be repeated. The presence of strong echo signal components in the signal can cause degraded or even faulty operation of the down-stream processing components, since these echo signal components may be mistaken for near-end speech. For these reasons, particular applications may set requirements for echo return loss enhancement (ERLE).

In the field of cellular radio communications, the typical (linear) echo cancellation arrangement described above has problems fulfilling the ERLE requirements, which are quite high for delayed systems such as the Global System for Mobile communications (GSM) and the Digital Advanced Mobile Phone System (DAMPS). The required ERLE figure can be as high as 40-50 dB in these cases. With good audio components it might be possible to achieve 30-35 dB ERLE by means of the linear echo canceler alone. To substantially eliminate the remainder, referred to as the residual echo, some extra processing is needed. Thus, an additional noise suppressor (NS) 123 may be provided in the system to handle more serious noise and echo components in the signal to be transmitted to the far-end user.

When the components that make up the hands-free arrangement are connected as an integral part of the overall device (e.g., telecommunications device or personal computer), the designers of such equipment can optimize various parameters to suit the particular type and arrangement of the hands-free components. For example, the respective gains of the microphone and loudspeaker amplifiers 119, 117 can be set to levels that are appropriate for the particular components selected for these functions. As another example, the echo canceling algorithm can be optimized (e.g., by choosing a suitable number of taps for the filter 101, an appropriate adaptation rate of the adaptation unit 103, and so on) to suit not only the particular microphone 105 and loudspeaker 109 to be used, but also to take account of their physical relationship with respect to one another and the type of environment (e.g., automobile passenger compartment, large/small office) into which they will be placed.

It is often the case, however, that the hands-free arrangement is connected as an accessory to the main device. For example, when a user of a handheld mobile telephone (the main device in this example) wishes to use the mobile telephone in an automobile, it is not uncommon for that user to connect the mobile telephone to automobile-mounted hands-free components by means of a suitable interface. In fact, for safety reasons many governments are now passing legislation prohibiting the use of anything other than hands-free equipment when the user wishes to communicate telephonically while operating an automobile. Hands-free adapters for existing mobile communication and other devices are thus proliferating. Under these circumstances, for any given make and model of mobile telephone or other device, the particular hands-free equipment owned and operated by the user may be any of a variety of makes and models, each with its own operating/performance characteristics. For example, some loudspeakers are more efficient than others, and therefore require, for a given acoustic power output, an input signal having a lower power level than is required by a less efficient speaker. Also, some loudspeakers generate more distortion than others, and therefore require different approaches to echo cancellation. It is similarly true that microphones differ from one another in characteristics such as sensitivity and directionality.

Further differences in operating characteristics arise from whether the hands-free components are always mounted in the same way with respect to one

another (e.g., whether the loudspeaker and microphone are always separated by a particular distance, and oriented in a particular direction with respect to one another) or whether this can vary. For example, in some vehicular adapters, the loudspeaker is fixed while the microphone may be one that is clipped to an article of clothing worn by the user. In such a case, the physical relationship between the loudspeaker and the microphone can vary greatly, and will certainly be unpredictable at the time that the user's main device (e.g., the mobile telephone) is being designed.

Thus, there is expected to be a wide variety of hands-free adapter kits for the user to choose from, ranging from low cost models intended to meet the demands of entry-level customers, up to the most expensive models that will provide high quality speech to both the near- and far-end users. In conventional systems, however, there is no mechanism provided for enabling the user's main equipment (e.g., the hand-held mobile phone) to distinguish one from another. As a consequence, this main equipment has to be designed assuming that the lowest quality hands-free adapter will be connected. When this is not the case, this design approach results in inefficient operation (e.g., echo cancellation algorithms that do more processing than necessary for the given equipment) and sub-optimal performance (e.g., operation with loudspeaker and microphone gain settings that are inappropriate for the actual devices being used).

U.S. Patent No. 5,036,532 discloses a portable telephone capable of discriminating between four different external accessories with power capabilities, one of which is a duplex hands-free adapter. When the telephone determines that a hands-free adapter is connected, it can change some of the settings of the telephone, such as the transmitter power level. No mention is made, however, of any capability for distinguishing between different hands-free adapters, each with its own operating characteristics. Thus, the telephone described by this reference is not capable of optimizing the performance of the telephone to suit the particular hands-free adapter being utilized.

U.S. Patent No. 5,349,698 discloses a radio communication apparatus that is capable of using a plurality of terminal units, and is capable of detecting what kind of terminal is being used. If the apparatus detects that a hands-free terminal is being used, it is capable of making corrections, such as volume adjustments. Again,

however, there is no mention of any capability for distinguishing between different hands-free adapters, each with its own operating characteristics. Thus, the telephone described by this reference is not capable of optimizing the performance of the telephone to suit the particular hands-free adapter being utilized.

German Patent Publication No. DE 197 15 596 A1 discloses an adapter for adapting between a mobile telephone and a hands-free unit. The adapter includes means for adjusting parameters, such as voltage levels, to be correct for different types of telephones. However, this document does not disclose any capability of distinguishing between different types of hands-free adapters, and making different adjustments as a result of this determination.

U.S. Patent No. 5,535,274 discloses a universal connection for a cellular telephone interface. The connection provides the capability to identify the type of telephone being used, and for downloading function-related parameters to a host assembly. The document fails to disclose, however, permitting the telephone to detect what type of hands-free adapter is connected, and making suitable adjustments based on that determination.

## SUMMARY

It is therefore an object of the present invention to provide methods and apparatuses that permit a main device to detect what type of hands-free equipment is connected.

It is another object of the present invention to provide methods and apparatuses that permit a main device to make adjustments to one or more operating parameters based on a determination of what type of hands-free equipment is connected.

The foregoing and other objects are achieved in methods and apparatuses for operating a main device. In one aspect of the invention, a main device receives self-identifying information from a hands-free adapter, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter.

In another aspect of the invention, the main device determines one or more settings as a function of the self-identifying information. Then, one or more components are adjusted in accordance with the one or more settings.

In yet another aspect of the invention, the self-identifying information may be make and model information. Alternatively, the self-identifying information may indicate one of a number of predefined performance classes, each of which is associated with a particular set of hands-free performance characteristics. In another alternative, the self-identifying information may indicate particular performance characteristics of the hands-free adapter.

In still another aspect of the invention, receiving self-identifying information from the hands-free adapter comprises retrieving the self-identifying information from one or more pins of a connector that links the apparatus and the hands-free adapter. Alternatively, receiving self-identifying information from the hands-free adapter may comprise receiving the self-identifying information from a wireless link between the apparatus and the hands-free adapter. In either case, this may further include establishing a serial communications link between the hands-free adapter and the apparatus; and receiving the self-identifying information via the serial communications link.

In yet another aspect of the invention, adjusting one or more components in accordance with the one or more settings comprises any one or more of the following: adjusting a gain of a microphone amplifier; adjusting a gain of a loudspeaker amplifier; adjusting one or more echo cancellation components in accordance with the one or more settings; adjusting a length of a filter that is used for echo cancellation; adjusting a filter adaptation rate of a filter that is used for echo cancellation; adjusting operation of a noise canceler in the apparatus; adjusting operation of a near-end voice detector in the apparatus; adjusting operation of a communication filter in the apparatus; setting the apparatus to operate in a full duplex mode; and setting the apparatus to operate in a semi-duplex mode.

In still another aspect of the invention, the hands-free adapter includes an interface and means for supplying self-identifying information to the interface. In some embodiments, the means for supplying self-identifying information to the interface may be a collection of one or more connector pins that are hard-wired to represent the self-

identifying information. In alternative embodiments, the means for supplying self-identifying information to the interface may be a programmable controller.

## BRIEF DESCRIPTION OF THE DRAWINGS

The objects and advantages of the invention will be understood by reading the following detailed description in conjunction with the drawings in which:

FIG. 1 is a block diagram of a conventional hands-free transceiver that includes an acoustic echo canceler in the form of an adaptive filter arrangement;

FIG. 2 is a block diagram of an exemplary embodiment of the invention; and

FIG. 3 is a flow chart depicting steps performed by a main device in accordance with one aspect of the invention.

## DETAILED DESCRIPTION

The various features of the invention will now be described with respect to the figures, in which like parts are identified with the same reference characters.

In accordance with one aspect of the invention, hands-free adaptors are provided with a capability of supplying self-identifying information to a main device, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter. The self-identifying information may be generated by including within the hands-free adaptor dedicated lines that are hardwired to predefined digital values (i.e., one's and zeroes) or alternatively by including within the hands-free adaptor a processor that generates the self-identifying information dynamically. The self-identifying information may be communicated to the main device by means of a physical connection, or via wireless means.

In accordance with other aspects of the invention, the main device uses the hands-free adaptor's self-identifying information to determine one or more settings to which elements of the main device are adjusted.

These and other aspects of the invention will now be described in greater detail with reference to an exemplary embodiment, a block diagram of which is depicted in FIG. 2. In the exemplary embodiment, the main device is a mobile telephone. In alternative embodiments, the main device could be a different type of

equipment, such as a personal computer running a communications program or even a speech recognition program. In these alternatives, the principles of receiving the hands-free adaptor's self-identifying information and making suitable adjustments to components of the main device are equally applicable, even if the particular components to be adjusted and the particulars of the adjustments are different from those typically found in a mobile telephone. Accordingly, the invention is not to be considered limited to the exemplary embodiments described herein.

Turning now to FIG. 2, a mobile telephone 203 is shown with a connector 213 that enables a bus 207 on the mobile telephone 203 to be coupled to a bus 205 located in a hands-free adapter 201. When not connected to the hands-free adapter 201, the mobile telephone 203 permits a near-end user to hear and speak to a far-end user by means of its own microphone and loudspeaker/earphone (not shown). These may be automatically disconnected using any of a number of well-known techniques, however, when the hands-free adapter 201 is connected to the connector 213. The hands-free adapter 201 includes a loudspeaker 109 for reproducing audio generated by a far-end user. The hands-free adapter 201 further includes a microphone 105 to convert the nearby acoustic sounds into a corresponding signal that may be processed and transmitted to the far-end user. The hands-free adapter 201 very commonly also includes a power amplifier 231 for the loudspeaker 109. This is a requirement in those embodiments that employ a wireless connection between the hands-free adapter 201 and the mobile telephone 203. The hands-free adapter 201 also very commonly includes a microphone preamplifier 229.

In order to generate a loudspeaker signal having an appropriate power level, the mobile telephone includes a speaker amplifier 217 having an adjustable gain. The mobile telephone 203 also includes a microphone amplifier 219 having an adjustable gain that permits a microphone signal to be generated at a suitable level, as will be described in greater detail below.

In order to address the acoustic echo problem described in the BACKGROUND section of this disclosure, the mobile telephone 203 further includes echo cancellation components including an adaptive filter 215, filter adaptation unit 221, and subtractor 107 arranged to generate an estimated echo signal 102 that is subtracted from a digital microphone signal 126, as described earlier. The mobile

telephone 203 may further include a noise suppression unit 223, although this is not required. In this embodiment, a near-end audio signal is supplied to a speech encoder 225, which generates an encoded speech signal that is supplied to other transmitter components (not shown) for transmission to the far-end user. The far-end signal (from which is derived a signal to be supplied to the loudspeaker) is received by receiver components (not shown) in the mobile telephone 203. The far-end signal is supplied to a speech decoder 227, which generates the digital input signal 112 which is processed as described earlier.

In accordance with one aspect of the invention, the hands-free adapter 201 supplies self-identifying information to the mobile telephone 203, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter. This information may take on many different forms, depending on the particular application. For example, the self-identifying information may be in the form of make and model identifying information. The mobile telephone 203 in this case needs to store a table so that it can look up the particular make and mode and determine what steps to take.

This approach has a drawback, however, in that if the mobile telephone's table is static, it may fail to recognize later-developed makes and models. In order to avoid this problem, and to avoid having to periodically update the mobile telephone's table, an alternative embodiment of the information may be used, in which the information includes detailed performance information about the particular microphone, loudspeaker and other components included in the hands-free adapter 201. An advantage of this approach is that the information can be as precise and detailed as desired. A disadvantage is that it may require a mechanism that permits a larger quantity of information to be communicated between the hands-free adapter 201 and the mobile telephone 203.

In yet another embodiment, the self-identifying information may identify one of a number of predefined performance classes of hands-free adapters. The performance classes may be predefined to describe a range of hands-free adapters of varying quality, ranging from very low-en models (usually of lower quality) to very high-end models (usually of very high quality). This approach avoids the problem of the mobile telephone 203 not recognizing a later-developed model of hands-free adapter

201, since even later developed models can be assigned a performance class that competently describes their performance characteristics. This approach further permits the pertinent information to be communicated between the hands-free adapter 201 and the mobile telephone 203 by means of very few bits (i.e., binary digits), since the information is essentially encoded.

In still other embodiments, combinations of the above types of self-identifying information can be used. For example, a hands-free adapter 201 may not be exactly described by any of the predefined performance classes. In such a case, a solution can be employed in which the hands-free adapter 201 communicates performance class information in combination with additional information. The communicated performance class may correspond to the performance class that most closely describes the hands-free adapter 201. The additional information can be just that information necessary to specify how the hands-free adapter 201 differs from the predefined performance class.

Just as the self-identifying information may take any of a number of forms, so too may alternative techniques be used for communicating the self-identifying information between the hands-free adapter 201 and the mobile telephone 203. In what is perhaps a simplest technique, one or more pins on the connector 213 may be dedicated to representing the information. Here, some pins would be tied to, say, a high voltage level (e.g., $V_{CC}$, or car battery voltage) in order to represent a binary '1', while other ones of the dedicated pins would be tied to ground for the purpose of representing a binary '0'. Together, the combination of '1's and '0's may represent, say, the performance class that best describes the hands-free adapter 201.

In an alternative embodiment, the hands-free adapter 201 may include a micro-controller 211 or other processor that generates the self-identifying information and transmits this to a mobile controller 209 or other processor in the mobile telephone 203. This communication is preferably performed by means of a serial communications link between the hands-free adapter 201 and the mobile telephone 203, although alternative techniques can also be used. The communications link itself can be by means of one or more dedicated lines on the connector 213. In alternative embodiments, however, a wireless connection is established between the hands-free adapter 201 and the mobile telephone 203, instead of the illustrated connector 213. A

wireless connection can be established and used in accordance with Bluetooth technology, which is a publicly available open specification for wireless communication of data and voice. Bluetooth specifications specify a system solution comprising hardware, software and interoperability requirements. Bluetooth equipment operates in a globally available 2.4 Ghz Industrial, Scientific and Medical (ISM) radiofrequency band, ensuring communication compatibility worldwide. Connections between Bluetooth-compatible devices are established almost-instantaneously, and are maintained even when devices are not within line of sight. The range of each radio is approximately 10 meters, but it can be extended to approximately 100 meters with an optional amplifier.

In one aspect of the invention, then, the hands-free adapter 201 provides self-identifying information to the mobile telephone 203. The self-identifying information is sufficient to permit the mobile telephone 203 to determine pertinent performance characteristics of the hands-free adapter 201. Another aspect of the invention relates to what the mobile telephone 203 does with this information. This aspect of the invention will be described with reference to the exemplary mobile telephone 203 depicted in FIG. 2 in conjunction with the flow chart depicted in FIG. 3. To facilitate an understanding of the invention, it will be assumed here that the hands-free adapter communicates a performance class identifier to the mobile controller 209 located in the mobile telephone 203. In other embodiments, however, the self-identifying information could take any of a number of alternative forms, as described earlier.

In step 301, the mobile controller 209 receives the hands-free adapter's self-identifying information in the form of a hands-free adapter performance class identifier. The mobile controller 209 uses this to optimize a number of settings in the mobile telephone 203 so that the most efficient settings can be used to attain the peak performance from this particular hands-free adapter 201.

In step 303, the mobile controller 209 adjusts the gain of the microphone amplifier 219 and of the loudspeaker amplifier 217. These settings, which are determined in advance for this particular performance class of hands-free equipment, ensure optimal performance with respect to saturation, interference and distortion associated with the microphone 105 and the loudspeaker 109.

In step 305, the length of the filter (i.e., the number of taps) is adjusted to best suit the performance class of hands-free equipment indicated by the hands-free adapter's self-identifying information. This step is performed out of recognition that lower quality hands-free equipment may require more filtering processing (i.e., more taps) than higher quality hands-free equipment. In order to avoid unnecessary echo-cancellation processing when less is required, thereby avoiding an unnecessary consumption of power (which is important, for example, when the mobile telephone has a wireless connection with the hands-free adapter 201, thereby eliminating the possibility of access to an external power supply) and longer processing time (including a longer time to achieve adaptive filter convergence which means a longer period of sub-optimal echo cancellation performance), the mobile controller 209 adjusts the filter algorithm so that it will do no more than what is required. For example, a performance class indicative of an office-based hands-free accessory will require a greater echo cancellation filter length than will be required for a performance class indicative of a headset. Depending on the particular embodiment, this filter length adjustment may comprise selecting or de-selecting physically implemented taps of a filter, or it may alternatively be an adjustment to one or more parameters stored in a memory, which parameters control the operation of a filtering program that is run on the mobile controller 209 or another processor located in the mobile telephone 203.

In step 307, the rate of filter adaptation (performed by the filter adaptation unit 221) is adjusted to suit the particular hands-free adapter 201. Again, this is performed because different hands-free adapter's have different echo characteristics. These differences arise not only from the quality differences between different kinds of microphones and different kinds of loudspeakers, but also because of installation distances that different hands-free adapter kits impose between microphone and loudspeaker, between microphone and mouth, and between loudspeaker and ear.

In step 309, other algorithms/components are adjusted to best suit the particular hands-free adapter 201 that is to operate with the mobile telephone 203. Such other algorithms/components include, but are not limited to, noise canceler, near-end voice detectors, other communication filters (uplink and downlink). The type of hands-free adapter 201 (which may always be associated with installation in a particular type of acoustic environment) may also cause the mobile telephone 203 to change from

a full duplex hands-free operation, in which the microphone 105 and loudspeaker 109 are always active at the same time, to semi-duplex hands-free operation, in which the loudspeaker 109 and microphone 105 take turns at being activated. In semi-duplex hands-free operation, the mobile controller 209 (or other controller/processor in the mobile telephone 203) may sense whether a far-end speech signal or a near-end speech signal has higher energy, and block transmission of the weaker signal.

In the above example, a number of mobile telephone settings were adjusted as a function of what type of hands-free adapter 201 was being employed. It will be recognized, however, that each of these exemplary settings need not always be adjusted in every embodiment. Rather, alternative embodiments may be devised in which only some of the exemplary adjustments are made. Furthermore, other types of adjustments, not illustrated here, may also be made as a function of the type of hands-free adapter 201 being employed.

The invention has been described with reference to particular embodiments. However, it will be readily apparent to those skilled in the art that it is possible to embody the invention in specific forms other than those of the preferred embodiments described above. This may be done without departing from the spirit of the invention.

For example, in the illustrated embodiments, the main device is a mobile telephone. However, this is not an essential feature of the invention. Rather, the invention can be practiced in any type of device that is capable of utilizing any of a variety of hands-free adapters.

It is apparent from the above that the quality difference between the least- and most-expensive hands-free adapters is extreme. Furthermore, the installation distances between microphone and loudspeaker, between microphone and mouth, and between loudspeaker and ear are also very different, depending on the particular accessory selected. In conventional main devices (e.g, mobile telephones), the particular type of hands-free equipment being utilized is unknown. Consequently, the amplifier gain, echo cancellation and other algorithms and components are designed to work with the worst case situation, which means the lowest quality hands-free equipment and the worst case installation environment. This often results in a negative influence on the performance of the high-end hands-free adapter.

The invention addresses these problems by permitting the main device to dynamically optimize internal settings to suit the particular hands-free adapter being employed.

In another aspect of the invention, a range of main devices (e.g., different models of mobile telephones) can be offered to the public, which main devices are associated with different qualities of performance. For example, a low-end mobile telephone can be designed to disable noise cancellation components unless it detects that it is connected to the highest-quality hands-free adapter 201. By contrast, a high-end mobile telephone can have the noise cancellation components always enabled, regardless of the type of hands-free adapter being employed.

Furthermore, it will be recognized that the invention can be practiced in conjunction with a wide variety of hands-free adapters 201. These include, but are not limited to: portable hands-free adapters (e.g., headsets), both wire and wireless; vehicle hands-free adapters, ranging from inexpensive easy-to-install kits up to more expensive fixed-mounted high-end kits; and office hands-free adapters, ranging from inexpensive desktop equipment to high-end conferencing equipment.

Thus, the preferred embodiments are merely illustrative and should not be considered restrictive in any way. The scope of the invention is given by the appended claims, rather than the preceding description, and all variations and equivalents which fall within the range of the claims are intended to be embraced therein.

WHAT IS CLAIMED IS:

1. A method of operating an apparatus, comprising the steps of:

receiving self-identifying information from a hands-free adapter, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter;

determining one or more settings as a function of the self-identifying information; and

adjusting one or more components in accordance with the one or more settings.

2. The method of claim 1, wherein the self-identifying information is make and model information.

3. The method of claim 1, wherein the self-identifying information indicates one of a number of predefined performance classes, each of which is associated with a particular set of hands-free performance characteristics.

4. The method of claim 1, wherein the self-identifying information indicates particular performance characteristics of the hands-free adapter.

5. The method of claim 1, wherein the step of receiving self-identifying information from the hands-free adapter comprises the step of retrieving the self-identifying information from one or more pins of a connector that links the apparatus and the hands-free adapter.

6. The method of claim 1, wherein the step of receiving self-identifying information from the hands-free adapter comprises the step of receiving the self-identifying information from a wireless link between the apparatus and the hands-free adapter.

7. The method of claim 1, wherein the step of receiving self-identifying information from the hands-free adapter comprises the steps of:

establishing a serial communications link between the hands-free adapter and the apparatus; and

receiving the self-identifying information via the serial communications link.

8. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting a gain of a microphone amplifier.

9. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting a gain of a loudspeaker amplifier.

10. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting one or more echo cancellation components in accordance with the one or more settings.

11. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting a length of a filter that is used for echo cancellation.

12. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting a filter adaptation rate of a filter that is used for echo cancellation.

13. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting operation of a noise canceler in the apparatus.

14. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting operation of a near-end voice detector in the apparatus.

15. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of adjusting operation of a communication filter in the apparatus.

16. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of setting the apparatus to operate in a full duplex mode.

17. The method of claim 1, wherein the step of adjusting one or more components in accordance with the one or more settings comprises the step of setting the apparatus to operate in a semi-duplex mode.

18. An apparatus comprising:

an input that receives self-identifying information from a hands-free adapter, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter; and

a controller that determines one or more settings as a function of the self-identifying information, and that adjusts one or more components in accordance with the one or more settings.

19. The apparatus of claim 18, wherein the self-identifying information is make and model information.

20. The apparatus of claim 18, wherein the self-identifying information indicates one of a number of predefined performance classes, each of which is associated with a particular set of hands-free performance characteristics.

21. The apparatus of claim 18, wherein the self-identifying information indicates particular performance characteristics of the hands-free adapter.

22. The apparatus of claim 18, wherein the input that receives self-identifying information from the hands-free adapter comprises one or more pins of a connector that links the apparatus and the hands-free adapter.

23. The apparatus of claim 18, wherein the input that receives self-identifying information from the hands-free adapter comprises radio elements that establish a wireless link between the apparatus and the hands-free adapter and that receive the self-identifying information from the wireless link.

24. The apparatus of claim 18, wherein the input that receives self-identifying information from the hands-free adapter comprises:

a communication element that establishes a serial communications link between the hands-free adapter and the apparatus; and

an input that receives the self-identifying information via the serial communications link.

25. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts a gain of a microphone amplifier.

26. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts a gain of a loudspeaker amplifier.

27. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts one or more echo cancellation components in accordance with the one or more settings.

28. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts a length of a filter that is used for echo cancellation.

29. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts a filter adaptation rate of a filter that is used for echo cancellation.

30. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts operation of a noise canceler in the apparatus.

31. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts operation of a near-end voice detector in the apparatus.

32. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that adjusts operation of a communication filter in the apparatus.

33. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that causes the apparatus to operate in a full duplex mode.

34. The apparatus of claim 18, wherein the controller that adjusts one or more components in accordance with the one or more settings comprises a controller that causes the apparatus to operate in a semi-duplex mode.

35. The apparatus of claim 18, wherein the apparatus is a telephone.

36. The apparatus of claim 35, wherein the telephone is a mobile telephone.

37. The apparatus of claim 35, wherein the one or more components include an acoustic echo canceler.

38. The apparatus of claim 18, wherein the apparatus is a general purpose digital computer.

39. A hands-free adapter comprising:

a microphone;

a loudspeaker;

an interface for coupling to a main device; and

means for supplying self-identifying information to the interface, wherein the self-identifying information distinguishes the hands-free adapter from at least one other hands-free adapter.

40. The hands-free adapter of claim 39, wherein the means for supplying self-identifying information to the interface is a collection of one or more connector pins that are hard-wired to represent the self-identifying information.

41. The hands-free adapter of claim 39, wherein the means for supplying self-identifying information to the interface is a programmable controller.

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[54]发明名称 对多种免提通信附件的自适应技术

[57]摘要

为了优化一个可以与多种各由一个具体性能质量表征的免提适配 器中任何一个免提适配器连接的诸如移动电话机之类的设备的性能，这个设备从所连接的具体免提适配器接收自标识信息。然后，这个设 备根据这自标识信息确定一个或多个设置，按照这些设置调整一个或 多个组件。例如，这样的调整可以包括拾音器增益调整、扬声器增益 调整和回声消除装置调整。

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# 权　利　要　求　书

1. 一种运行一个设备的方法，所述方法包括下列步骤:

接收来自一个免提适配器的自标识信息，其中所述自标识信息将所述免提适配器与至少一种其他免提适配器区分开来;

根据所述自标识信息确定一个或多个设置；以及

按照所述一个或多个设置调整一个或多个组件.

2. 权利要求1的方法，其中所述自标识信息是产品和型号信息.

3. 权利要求1的方法，其中所述自标识信息表明若干个各与一组具体的免提工作特性有关的预定性能类别中的一个类别.

4. 权利要求1的方法，其中所述自标识信息表明所述免提适配器的具体性能特性.

5. 权利要求1的方法，其中所述接收来自免提适配器的自标识信息的步骤包括从一个链接所述设备和所述免提适配器的连接器的一个或多个插脚提取所述自标识信息的步骤.

6. 权利要求1的方法，其中所述接收来自免提适配器的自标识信息的步骤包括从一个在所述设备和所述免提适配器之间的无线链路接收所述自标识信息的步骤.

7. 权利要求1的方法，其中所述接收来自免提适配器的自标识信息的步骤包括下列步骤:

在所述免提适配器与所述设备之间建立一个串行通信链路；以及

通过所述串行通信链路接收所述自标识信息.

8. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整一个拾音器放大器的增益的步骤.

9. 权利要求1的方法，　其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整一个扬声器放大器的增益的步骤.

10. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括按照所述一个或多个设置调整一个或多个回声消除组件的步骤.

11. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整一个用来进行回声消除的滤波器的长度的步骤.

12. 权利要求1的方法，其中所述按照一个或多个设置调整一个或

多个组件的步骤包括调整一个用来进行回声消除的滤波器的滤波器自适应速率的步骤.

13. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整所述设备内的一个噪声消除器的操作的步骤.

14. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整所述设备内的一个近端语音检测器的操作的步骤.

15. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括调整所述设备内的一个通信滤波器的操作的步骤.

16. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括将所述设备设置成以全双工方式工作的步骤.

17. 权利要求1的方法，其中所述按照一个或多个设置调整一个或多个组件的步骤包括将所述设备设置成以半双工方式工作的步骤.

18. 一种设备，所述设备包括:

一个接收来自一个免提适配器的自标识信息的输入端，其中所述自标识信息将所述免提适配器与至少一种其他免提适配器区分开来;以及

一个根据所述自标识信息确定一个或多个设置、按照所述一个或多个设置调整一个或多个组件的控制器.

19. 权利要求18的设备，其中所述自标识信息是产品和型号信息.

20. 权利要求18的设备，其中所述自标识信息表明若干个各与一组具体的免提工作特性有关的预定性能类别中的一个类别.

21. 权利要求18的设备，其中所述自标识信息表明所述免提适配器的具体性能特性.

22. 权利要求18的设备，其中所述接收来自免提适配器的自标识信息的输入端包括一个链接所述设备和所述免提适配器的连接器的一个或多个插脚.

23. 权利要求18的设备，其中所述接收来自免提适配器的自标识信息的输入端包括在所述设备与所述免提适配器之间建立一个无线链

路的无线电构件和从所述无线链路接收所述自标识信息的无线电构件.

24. 权利要求18的设备，其中所述接收来自免提适配器的自标识信息的输入端包括:

在所述免提适配器与所述设备之间建立一个串行通信链路的通信构件; 以及

一个通过所述串行通信链路接收所述自标识信息的输入端.

25. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整一个拾音器放大器的增益的控制器.

26. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整一个扬声器放大器的增益的控制器.

27. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个按照所述一个或多个设置调整一个或多个回声消除组件的控制器.

28. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整一个用来进行回声消除的滤波器的长度的控制器.

29. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整一个用来进行回声消除的滤波器的滤波器自适应速率的控制器.

30. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整所述设备内的一个噪声消除器的操作的控制器.

31. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整所述设备内的一个近端语音检测器的操作的控制器.

32. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个调整所述设备内的一个通信滤波器的操作的控制器.

33. 权利要求18的设备，其中所述按照一个或多个设置调整一个

或多个组件的控制器包括一个使所述设备以全双工方式工作的控制器.

34. 权利要求18的设备，其中所述按照一个或多个设置调整一个或多个组件的控制器包括一个使所述设备以半双工方式工作的控制器.

35. 权利要求18的设备，其中所述设备是一个电话机.

36. 权利要求35的设备，其中所述电话机是一个移动电话机.

37. 权利要求35的设备，其中所述一个或多个组件包括一个回声消除器.

38. 权利要求18的设备，其中所述设备是一个通用数字计算机.

39. 一种免提适配器，所述免提适配器包括:

一个拾音器;

一个扬声器;

一个与一个主设备连接的接口; 以及

为接口提供自标识信息的装置，其中所述自标识信息将所述免提适配器与至少一种其他免提适配器区分开来.

40. 权利要求39的免提适配器，其中所述向接口提供自标识信息的装置是一个由一个或多个硬线连接成表示所述自标识信息的连接器插脚组成的集体.

41. 权利要求39的免提适配器，其中所述向接口提供自标识信息的装置是一个可编程控制器.

# 说　明　书

## 对多种免提通信附件的自适应技术

### 发明背景

本发明与电信及其他音频设备有关，具体地说，与使用与电信及其他音频设备配合的免提附件的技术有关.

在诸如具有对讲机之类的电信领域内，以及在蜂窝电话通信中，通常所希望的是允许用户操作通信设备而不需要一直占着用户的手.这在诸如汽车之类的环境中可能是很重要的，驾驶员握着电话设备可能不仅危及他的安全而且还危及路上其他人的安全.自由地使用一个人的手去做除了握着拾音器之外的某些事情在诸如个人计算机互联网通信、计算机语音识别和视听表现系统之类的其他应用情况下也是有用的.

为了适应这些重要的需要，业已开发了所谓的“免提”装置，使拾音器和扬声器可以安装或者放置在免提环境内，从而不需要握住它们.例如，在汽车应用的情况下，蜂窝电话的拾音器可以安装在遮阳板上，而扬声器可以是一个安装在仪表板上的单元，也可以是与汽车立体声设备关联的一个单元.采用这样安装的组件，蜂窝电话用户可以继续交谈而不需握住蜂窝单元或者它的手机.类似，个人计算机通常具有例如安装在监视器内相互比较接近的拾音器和扬声器.

免提配置的一个问题是拾音器除了免提装置的用户（所谓“近端用户”）的语音之外还会捡取来自附近扬声器的声音.这也是一些诸如越来越小的手持移动电话机之类的非免提设备中的问题.（由于尺寸小，移动电话机的拾音器不能完全与它的扬声器发出的声音隔离.）拾音器检测到由扬声器产生的声音会在许多种应用中引起问题.例如，在通信设备中，通信系统整体引入的延迟可能使来自扬声器的声音被通话的另一端（所谓的“远端”）听到，成为他自己的声音的回声.这样的回声降低了声音的质量，减轻回声是所希望的，甚至是绝对必要的，因为存在延迟回声严重地妨碍了远端用户的进行正常交谈的能力.类似的问题也可能存在于例如一些通过扬声器合成语音和包括对拾音器检测到的所说出的命令或其他词进行识别和响应的语音识别组件的自动化系统.在这样的应用中，在拾音器信号中存在合成语音的

回声可能会严重地降低语音识别组件的性能．减轻这样的回声的措施包括利用自适应回声消除滤波器、回声衰减器，或者这两者的某种组合．

作为通常免提装置的有代表性的例子，图1示出了一个具有传统的呈自适应滤波器配置的回声消除器的典型“免提”移动电话机．免提通信环境可以例如是设置移动电话机的汽车内部．这样一个环境可能会影响声信号的传播，而这影响通常是未知的．下面，这类环境将在本说明书中标为未知系统H(z)．免提移动电话机包括一个扬声器109，为近端用户提供远端声信号．免提移动电话机还包括一个拾音器105，将近端声信号（通常由近端用户产生）变换成电或其他信号，传输给远端用户．

在工作中，表示比方说远端用户声音的数字输入音频信号112提供给数模(D/A)变换器115．所得到的模拟信号由扬声器放大器117适当放大后加到扬声器109上．在反方向上，拾音器105接收声能，将它变换成模拟信号．这信号由拾音器放大器119适当放大后，由模数变换器121变换成一系列数字样点．（人们意到，在这个典型的实施例中，大部分处理都是对数字信号进行的，因此需要D/A和A/D变换器115、121．然而，在其他实施例及其他应用中，在这里所示出的所有处理都可以对模拟信号进行．在这样的实施例中，就可以不需要D/A和A/D变换器115、121．）然后，所得到的信号126受到进一步的处理，以便最终传输给远端用户．

在这样的配置中，拾音器105用来检测用户的语音，但是可能受到扬声器109发出的声音信号的不希望有的影响．就是这种不希望有的影响使回声信号进入了系统．

减小甚至消除回声的电路包括一个诸如自适应有限脉冲响应（FIR）或无限脉冲响应（IIR）滤波器之类的自适应滤波器101、一个诸如最小均方（LMS）互相关器之类的自适应单元103和一个减法器107．如果假定拾音器105接收到的声音只是来自扬声器109（当然，实际上拾音器还接收来自预定用户的声音和背景噪声），然后产生数字化的回声信号126（也标为“u”）．在工作中，自适应滤波器101产生一个回声估计信号102，通常标为信号$\hat{u}$．这个回声估计信号102可以作为远端信号112与滤波器101的m个滤波器加权系数（$h_i$）的序列的卷积（见

式1）.

$$\hat{u}(n)=\sum_{i=0}^{1-n} h_i \cdot x(n-i) \qquad (1)$$

其中：

$x(n)$为输入信号，

$m$为加权系数的个数，以及

$n$为样点数.

在这些加权系数正确设置时，自适应滤波器101产生一个近似等于扬声器109在未知系统H(z)内产生的响应的脉冲响应．从输入的数字化回声信号126中减去自适应滤波器101产生的回声估计信号102，产生一个经回声滤波的拾音器信号，这个信号也称为误差信号$e(n)$，因为它的能量在到达拾音器105的声音只是由扬声器109发出的声音时完全取决于回声估计信号102的误差：

$$e(n)=u(n)-\hat{u}(n) \qquad (2)$$

在理论上，可以通过减去回声估计信号102从数字化的回声信号126中除去任何由扬声器109引入的未知系统H(z)的回声响应．因此，在数字化的回声信号126还包括可归因于预定近端用户的信号分量时，所得到的差信号理论上应该只包括近端用户信号分量．通常，有效地消除回声所需的加权系数（此后就称为“系数”）的个数将取决于应用情况．对于手持电话机来说，系数少于一百个可能是适当的．对于在汽车内的免提电话机，系数大约需要200至400个．一个大房间可能需要一个有1000个以上的系数的滤波器，以便提供适当的回声消除.

可以看出，回声消除器的有效性直接与自适应滤波器101能复现未知系统H(z)的脉冲响应的程度有关．于是，这直接与滤波器101所用的这组系数$h_i$有关.

有利的是提供一种机制，可以动态地改变这些系数$h_i$，使自适应滤波器101适应未知系统H(z)的改变．在一个具有免提蜂窝装置的汽车内，在车窗或车门打开或关闭时就可能出现这样的改变．实际上，在

这种及其他环境中，未知系统H(z)的改变还可以由许多环境方面的各种改变引起，诸如人员或物体的移动，或者拾音器105和扬声器109相互之间的相对位置或朝向的改变。一种众所周知的系数自适应系统是最小均方（LMS）过程，它首先是由Widrow和Hoff在1960年引入的，由于它的性能有效和稳健而经常采用。在应用于回声消除问题时，LMS过程是一种随机梯度步长方法，利用梯度的粗（有噪声的）估计$\underline{g}(n)=e(n)\cdot\underline{x}(n-1)$安排一个朝使误差信号的能量（和均方误差）e(n)最小的增量步长，其中$\underline{x}(n)$是向量记法，相应于表达式$\underline{x}(n)=[x(n)\ x(n-1)\ x(n-2)\ldots\ x(n-m+1)]$。LMS过程$e(n)\cdot\underline{x}(n)$产生的更新信息用来确定下一个样点的系数的值。计算下一个系数值$h_i(n+1)$的表达式为：

$$h_\Phi(n+1)=h_\Phi(n)+\mu e(n)\cdot x(n-i),\qquad i=0..m-1 \qquad (3)$$

其中：

$x(n)$为数字化输入信号112的第n个样点，

$h_i$为滤波器加权系数，

$i$标示特定的系数，

$m$为加权系数的个数，

$n$为样点号码，以及

$\mu$为步长或更新增益参数。

LMS方法由于有效和简单，非常普遍地用于一些实时应用。它产生增长部分的信息，每个增长部分可以具有正值或负值。LMS过程产生的信息可以提供给自适应滤波器101，更新它的系数。

再来看图1，这个典型的传统回声消除电路包括一呈LMS互相关器的滤波器自适应单元103，用来为滤波器101提供系数更新信息104。在这种配置中，滤波器自适应单元103监视表示数字化回声信号126减去滤波器101产生的回声估计信号102的误差信号$e(n)$。回声估计信号102如上所述利用滤波器自适应单元103提供给自适应滤波器101的更新信息104产生。自适应滤波器101的这些系数$h_i$按式（3）所示累加更新信息104。

无论是否存在诸如如上所述的回声消除电路，所产生的要进一步处理（例如，传输给通信系统的远端用户，或者进行近端语音识别或

控制回声消除滤波器101的操作）的信号经常还可能包括回声和/或其他噪声分量．这是可能发生的，例如因为自适应滤波器还没有收敛到一个完全适应状态，甚至在这种收敛后每当未知环境H(z)有改变时，仍需要重复这个自适应过程．在信号中存在强回声信号分量可能引起降低下游处理组件的性能，甚至导致下游处理组件的误操作，因为这些回声信号分量可能被误认为近端的语音．为此，对于具体的应用可以规定一些对于增强回声返回损耗（ERLE）的要求．

在蜂窝无线电通信领域内，如上所述的典型（线性）回声消除配置具有要满足ERLE要求的问题，这些要求对于诸如全球移动通信系统（GSM）和高级数字移动式电话系统（DAMPS）之类的延迟系统来说是相当高的．在这种情况下，所要求的ERLE指标高达40-50dB．采用良好的音频组件，单通过线性回声消除器可以达到30-35dB的ERLE．为了基本上消除称为剩余回声的残余成分，需要一些额外的处理．因此，在系统内可以配置一个附加的噪声抑制器（NS）123，用来处理在要发送给远端用户的信号中的比较严重的噪声和回声成分．

在组成免提配置的这些组件连接成整个设备（例如，电信设备或个人计算机）的不可分割的部分时，这种设备的设计者可以将各个参数优化成适合免提组件的具体类型和配置．例如，可以将拾音器和扬声器放大器119、117的相应增益设置成适合于为这些功能所选定的具体组件．作为另一个例子，可以将回声消除算法优化（例如，通过选择适当的滤波器101的抽头数、适当的自适应单元103的自适应速率等等）成不仅适合所用的具体拾音器105和扬声器109，而且考虑到它们之间的相对物理关系和它们所要放到的环境的类型（例如，汽车、客舱、大/小办公室）．

然而，通常的情况是免提装置作为一个附件连接到主设备上．例如，在一个手持移动电话机（在这个例子中是主设备）的用户希望在汽车内使用移动电话机时，用户经常是通过一个适当的接口将移动电话机连接到安装在汽车上的一些免提组件上．实际上，为了安全起见，许多政府现在都正通过立法禁止在驾驶汽车时使用免提装置之外的任何设备进行电话通信．因此，对于现有的移动通信及其他设备的免提适配器的需要激增．在这种情况下，对于任何给定的移动电话机或其他设备的产品和型号，用户拥有和操作的具体免提装置可能是各有各

的工作/性能特性的多种产品和型号中的任何一个．例如，有些扬声器效率比其他一些高，因此对于给定的输出声功率，要求输入信号具有比效率较低的扬声器所需的低的功率．此外，有些扬声器产生比其他一些大的失真，因此需要用不同的方法进行回声消除．对于在诸如灵敏度和方向性之类的特性上相互不同的一些拾音器来说情况与此类似．

免提组件是否始终以相对相同的方式安装（例如，是否扬声器与拾音器始终相对隔开一段特定的距离和朝着一个特定的方向）或者是否可以改变都会进一步引起工作特性的差别．例如，在有些车辆适配器中，扬声器是固定的，而拾音器可能夹在用户所穿着的衣服上．在这种情况下，扬声器与拾音器之间的物理关系可能改变很大，这在设计用户的主设备（例如，移动电话机）的时候当然是无法预测的．

因此，可以设想，有各式各样的免提适配器套件，从满足初级客户需要的低价型号直到为近端和远端用户提供高质量语音的高价型号，供用户选择．然而，在一些传统系统中，还没有一种机制使用户的主设备（例如，手持移动电话机）区分这些．因此，主设备必须在要与低档免提适配器连接的假设情况下进行设计．在情况不是这样时，这种设计方法导致低效工作（例如，回声消除算法进行比给定设备所需的多的处理）和非最佳性能（例如，以对于所用的实际设备不适当的扬声器和拾音器的增益设置进行工作）．

美国专利No. 5,036,532揭示了一种能够区别四种不同的具有驱动能力的外部辅助装置，其中之一是双工免提适配器．在电话机确定连接的是一个免提适配器时，它可以改变电话机的一些设置，例如发射机功率．然而，其中并没有提到任何区别各有各的工作特性的不同免提适配器的能力．因此，这个专利所揭示的电话机不能将电话机的性能优化成适合所用的具体免提适配器．

美国专利No. 5,349,698揭示了一种能用多种终端单元并能检测正在用的是哪种终端的无线电通信设备．如果这种设备检测到正在使用的是一个免提终端，它就能进行校正，例如作音量调整．然而，其中同样也没有提到有可能区别各有各的工作特性的不同免提适配器．因此，这个专利所揭示的电话机不能将电话机的性能优化成适合所用的具体免提适配器．

德国专利刊物No. DE 197 15 596 A1揭示了一种在移动电话机和免提单元之间进行适配的适配器. 这种适配器包括为不同类型的电话机将诸如电压之类的参数调整到正确的装置. 然而, 这个文件并没有揭示任何区别不同类型的免提适配器和按确定结果进行不同调整的能力.

美国专利No. 5,535,274揭示了一种对于蜂窝电话接口的通用连接. 这种连接为主机提供了识别在用的电话机的类型和下载与功能有关的参数的能力. 然而, 这个文件并没有揭示允许电话机检测所连接的是什么类型的免提适配器和根据所确定的类型进行适当的调整.

## 发明概要

因此, 本发明的目的是提供允许主设备检测所连接的是什么类型的免提装置的方法和装置.

本发明的另一个目的是提供允许主设备根据确定所连接的是什么类型的免提装置, 对一个或多个工作参数进行调节的方法和装置.

上述及其他一些目的是用对主设备进行控制的方法和装置达到的. 就本发明的一种情况来说, 主设备接收来自免提适配器的自标识信息, 这种自标识信息将本免提适配器与至少一种其他免提适配器区分开来.

在本发明的另一种情况下, 主设备根据自标识信息确定一个或多个设置. 然后, 按照这一个或多个设置调整一个或多个组件.

在本发明的另一种情况下, 自标识信息可以是产品和型号信息. 或者, 自标识信息也可以表明是若干个各与一组具体的免提工作特性有关的预定性能类别之一. 在另一个供选择的方案中, 自标识信息可以表明免提适配器的具体工作特性.

在本发明的又一种情况下, 接收来自免提适配器的自标识信息包括从一个将设备与免提适配器链接的连接器的一个或多个引线提取自标识信息. 或者, 接收来自免提适配器的自标识信息也可以包括从在设备与免提适配器之间的无线链路接收自标识信息. 但不论是哪种情况, 这还可以包括: 在免提适配器与设备之间建立一个串行通信链路; 以及通过这个串行通信链路接收自标识信息.

在本发明的又一种情况下, 按照一个或多个设置调整一个或多个组件包括以下任何一个或多个步骤: 调整一个拾音器放大器的增益;

调整一个扬声器放大器的增益；按照一个或多个设置调整一个或多个回声消除组件；调整一个用来进行回声消除的滤波器的长度；调整一个用来进行回声消除的滤波器的滤波器自适应速率；调整设备内一个噪声消除器的操作；调整设备内一个近端语音检测器的操作；调整设备内一个通信滤波器的操作；将设备设置成以全双工方式进行工作；以及将设备设置成以半双工方式进行工作.

在本发明的还有一种情况下，免提适配器包括一个接口和为接口提供自标识信息的装置．在一些实施例中，为接口提供自标识信息的装置可以是一个由一个或多个硬接线成表示自标识信息的接线插脚组成的集体．在另一些实施例中，为接口提供自标识信息的装置可以是一个可编程控制器.

## 附图简要说明

从以下结合附图所作的详细说明中可以更清楚地看到本发明的这些目的和优点，在这些附图中:

图1为一个传统的包括呈自适应滤波器配置的回声消除器的免提收发信机的方框图;

图2为本发明的一个典型实施例的方框图；以及

图3为示出按照本发明的一种情况由主设备执行的一些步骤的流程图.

## 详细说明

下面结合这些附图说明本发明的各个特色部分，在这些附图中同样的部分标以同样的标识字符.

就本发明的一种情况来说，免提适配器配置有为主设备提供自标识信息的能力，这种自标识信息将本免提适配器与至少一种其他免提适配器区分开来．可以通过在免提适配器内包括一些硬连接到一些预定的数字值（即一些“1”和“0”）的专用线或者通过在免提适配器内配置一个动态地产生自标识信息的处理器来产生自标识信息．自标识信息可以通过物理连接或无线装置传送给主设备.

按照本发明的其他一些情况，主设备利用免提适配器的自标识信息来确定一个或多个设置，调整主设备的一些构件.

下面将结合方框图如图2所示的典型实施例详细说明本发明的这些及其他一些情况．在这个典型实施例中，主设备是一个移动电话机.

在另一些实施例中，主设备可以是一个不同类型的设备，例如是一个运行通信程序甚至语音识别程序的个人计算机．在这些可能的情况中，同样可用接收免提适配器的自标识信息和对主设备的组件进行适当调整的原理，虽然需调整的具体组件和进行调整的具体情况与通常在移动电话机内所看到的不同．因此，本发明并不局限于在这里所说明的这个典型实施例．

现在来看图2，所示的移动电话机203有一个连接器213，它使移动电话机203的总线207可以与免提适配器201内的总线205连接．在没有与免提适配器201连接时，移动电话机203允许近端用户通过它自有的拾音器和扬声器/耳机（未示出）与远端用户对话．然而在免提适配器201连接到连接器213上时，可以采用任何众所周知的技术自动地使它们断开．免提适配器201包括一个再现远端用户产生的声音的扬声器109．免提适配器201还包括一个将附近的声音变换成可以在处理后发送给远端用户的相应信号的拾音器105．免提适配器201通常还包括一个与扬声器109配合的功率放大器231．这是在那些在免提适配器201与移动电话机203之间采用无线连接的实施例中的一个要求．免提适配器201通常还包括一个拾音器前置放大器229．

为了产生一个功率适当的扬声器信号，移动电话机包括一个增益可调的扬声器放大器217．移动电话机203还包括一个增益可调的拾音器放大器219，用来将拾音器信号放大到一个适当的电平，这在下面还要详细说明．

为了考虑在本说明书的发明背景部分所述的回声问题，移动电话机203还包括一些回声消除组件，这些回声消除组件有自适应滤波器215、滤波器自适应单元221和减法器107，配置成产生一个要从数字拾音器信号126中减去的回声估计信号102，如前面所说明的那样．移动电话机203还可以包括一个噪声抑制单元223，虽然这不是一定需要的．在这个实施例中，近端音频信号提供给语音编码器225，由它产生一个编码语音信号，提供给其他发射机组件（未示出），以便传输给远端用户．远端信号（从中得出要提供给扬声器的信号）由移动电话机203内接收机组件（未示出）接收．远端信号提供给语音解码器227，由它产生要处理的数字输入信号112，如前面所说明的那样．

就本发明的一种情况来说，免提适配器201向移动电话机203提供

自标识信息，这种自标识信息将本免提适配器与至少一种其他免提适配器区分开来．这种信息可以呈现为许多不同的形式，取决于具体应用．例如，自标识信息可以呈产品和型号标识信息．在这种情况下，移动电话机203需要存储一个表，使它可以查找具体的产品和型号，确定要采取什么措施．

然而，这样处理有一个缺点，因为如果移动电话机的表是不变的，它就不能识别以后开发的产品和型号．为了避免这个问题和避免必须周期性地更新移动电话机的表，可以采用另一种信息，这种信息包括有关免提适配器201内的具体拾音器、扬声器及其他组件的详细性能信息．这样的优点是信息可以精确和详细到如所要求的那样．缺点是它需要一种机制，可以在免提适配器201与移动电话机203之间传送大量信息．

在又一个实施例中，自标识信息可以标识免提适配器的若干个预定性能类别中的一个类别．可以将这些性能类别预先规定成揭示各种质量不同的免提适配器，从很低档型号（通常具有较低质量）到很高档型号（通常具有很高的质量）．这样处理避免了移动电话机203不能识别以后开发的免提适配器201的型号的问题，因为即使是以后开发的型号，也可以为它们指定一个能揭示它们的工作特性的性能类别．这样处理还使有关信息在免提适配器201与移动电话机203之间能用很少的比特（即二进制数字）传送，因为这信息实质上是编码的．

在其他一些实施例中，可以采用以上类型的自标识信息的一些组合．例如，可能一个免提适配器201不能由任何预定的性能类别确切表达．在这种情况下，可以采用一种解决方案，免提适配器201将性能类别信息与附加信息组合在一起传送．所传送的性能类别可以与最接近免提适配器201的性能的性能类别相应．附加信息可以只是说明免提适配器201与这个预定的性能类别有多少不同的信息．

恰如自标识信息可以采取若干形式中的任何一种形式那样，也可以采用其他技术在免提适配器201与移动电话机203之间传送自标识信息．在一种也许是最简单的技术中，可以将连接器213上的一个或多个插脚供表示这信息专用．此时，一些专用插脚要接到比方说一个高电压电平（例如，VCC或者汽车电池电压）上以表示二进制“1”，而另一些要接到地线上以表示二进制“0”．这些“1”和“0”组合在一起

可以表示比方说能最好地揭示这个免提适配器201的性能类别.

在另一个实施例中，免提适配器201可以包括一个微控制器211或其他处理器，用来产生自标识信息和将它发送给移动电话机203内的移动台控制器209或其他处理器．这样的通信最好是通过免提适配器201与移动电话机203之间的串行通信链路进行，当然也可以采用其他技术．这个通信链路本身可以是连接器213上的一根或多根专用线．然而，在另一些实施例中，在免提适配器201与移动电话机203之间建立起一个无线连接来代替所例示的连接器213．可以按照蓝牙技术建立和利用无线连接，蓝牙技术是一种对于数据和语音无线通信的公众可用的开放规范．蓝牙规范规定了一种系统解决方案，包括对硬件、软件和互动性的要求．蓝牙设备工作在全球可用的2.4 Ghz工业、科学和医疗（ISM）无线电频带，从而保证了在世界范围内的通信兼容性．蓝牙兼容设备之间的连接几乎瞬时建立，即使在设备不在视线内时也可运用．每个无线电设备的作用距离大约为10米，但是采用一个可选放大器后可以扩展到大约100米．

在本发明的一种情况下，免提适配器201为移动电话机203提供自标识信息．自标识信息足以使移动电话机203可以确定免提适配器201有关的工作特性．本发明的另一种情况与移动电话机203用这信息做些什么有关．本发明的这种情况将结合图3所示的流程图参考图2所示的典型移动电话机203进行说明．为了便于理解本发明，在这里将假设免提适配器将一个性能类别标识符传送给移动电话机203内的移动台控制器209．然而，在其他实施例中，自标识信息可以采取若干其他形式中的任何一种形式，如前面所说明的那样．

在步骤301，移动台控制器209接收到免提适配器的呈免提适配器性能类别标识符的自标识信息．移动台控制器209用此优化移动电话机203内的若干设置，这样就可以根据这个具体的免提适配器201用这些最有效的设置达到最佳性能．

在步骤303，移动台控制器209调整拾音器放大器219和扬声器放大器217的增益．这些设置对于免提设备的这个具体的性能类别是事先确定的，可以保证对于与拾音器105和扬声器109关联的饱和、干扰和失真来说是最佳的性能．

在步骤305，将滤波器的长度（即抽头数）调整到最适合由免提适

配器的自标识信息表示的免提设备的性能类别．执行这个步骤是由于认识到低质量的免提设备可能需要比高质量的免提设备多的滤波处理（即，更多的抽头）．为了在不很需要时避免不必要的回声消除处理从而避免不必要的功率消耗（这例如在移动电话机与免提适配器201无线连接时是重要的，因此可以不用接入外部电源）和较长的处理时间（包括达到自适应滤波器收敛的时间较长，这意味着回声消除性能不是最佳的时间较长），移动台控制器209调整滤波算法，使它只做需要做的处理．例如，一个表示基于办公室的免提辅助装置的性能类别将需要比一个表示耳机的性能类别所需要的长的回声消除滤波器长度．根据具体的实施例，滤波器长度调整可以包括选择或重选物理实现的滤波器抽头，或者也可以是对存储在一个存储器内的一个或多个参数的调整，这些参数控制一个在移动台控制器209上或者在移动电话机203内的另一个处理器上运行的滤波程序的操作．

在步骤307，将滤波器自适应速率调整（由滤波器自适应单元221执行）到适合具体的免提适配器201．同样，执行这个步骤是因为不同的免提适配器具有不同的回声特性．这些差别不仅是由于在各种拾音器和各种扬声器之间的质量差别引起的，而且还由于不同的免提适配器套件的拾音器与扬声器之间、拾音器与嘴之间和扬声器与耳朵之间的安装距离的差别引起．

在步骤309，将其他算法/组件调整到最适合与移动电话机203一起工作的具体免提适配器201．这些其他算法/组件包括（但并不局限于）噪声消除器、近端语音检测器和其他通信滤波器（上行链路和下行链路滤波器）．免提适配器201的类型（这可能始终与在具体类型的声学环境内的安装情况有关）也可以使移动电话机203从全双工免提工作（拾音器105和扬声器109始终同时激活）改变为半双工免提工作（扬声器109和拾音器105轮流激活）．在半双工免提工作中，移动台控制器209（或者移动电话机203内的其他控制器/处理器）可以检测是远端语音信号还是近端语音信号具有较高的能量，从而闭塞较弱的那个信号的传输．

在以上这个例子中，根据正在使用的免提适配器201的类型调整若干移动电话机设置．然而，可以看到，这些典型设置没有必要每个在所有的实施例中都要调整．确切些说，可以设计出另一些实施例，其中只进行所述典型调整中的一些调整．此外，根据所用的免提适配器201

的类型也可以进行在这里没有例示的其他类型的调整.

以上结合具体实施例对本发明作了说明. 然而, 熟悉该技术领域的人员很容易看到, 本发明可以用上面所说明的这些优选实施例的形式之外的一些具体形式来体现. 这可以根据本发明的精神实现.

例如, 在所例示的这些实施例中, 主设备是一个移动电话机. 然而, 这并不是本发明的根本特色. 本发明可以用于任何类型的能运用任何免提适配器的设备.

由上可见, 最低与最高档的免提适配器之间的质量差别是极大的. 此外, 拾音器与扬声器之间、拾音器与嘴之间和扬声器与耳朵之间的安装距离也很不同, 这取决于所选的具体辅助装置. 在传统的主设备(例如, 移动电话机)中, 要用的免提装置的具体类型是未知的. 因此, 放大器增益、回声消除和其他算法和组件都设计成能在最坏的情况(这意味着质量最低的免提装置和安装环境最差的情况)下工作. 这通常导致对高档免提适配器的性能有相反的不良影响.

本发明通过使主设备可以动态地将内部设置优化成适合所用的具体免提适配器考虑了这些问题.

在本发明的另一种情况下, 可以向公众提供各种不同性能质量的主设备(例如, 不同型号的移动电话机). 例如, 低档的移动电话机可以设计成能停用噪声消除组件, 除非它检测到它所连接的是高质量的免提适配器201. 相反, 高档移动电话机可以始终使噪声消除组件启用, 而无论所用的是哪种免提适配器.

此外, 可以看到, 本发明可以用于各种免提适配器201. 包括(但不局限于): 便携式免提适配器(例如, 头机), 有线的和无线的; 车辆免提适配器, 从安装容易的经济套件到固定安装的昂贵高档套件; 以及办公室免提适配器, 从经济的桌面装置到高档的会议装置.

因此, 这些优选实施例只是例示性的, 无论如何不应该认为是限制性的. 本发明的专利保护范围由所附权利要求书给出而不是前面的说明, 在这些权利要求的范围内的所有变型和等效实现方式都应列入本发明的专利保护范围.

# 说 明 书 附 图

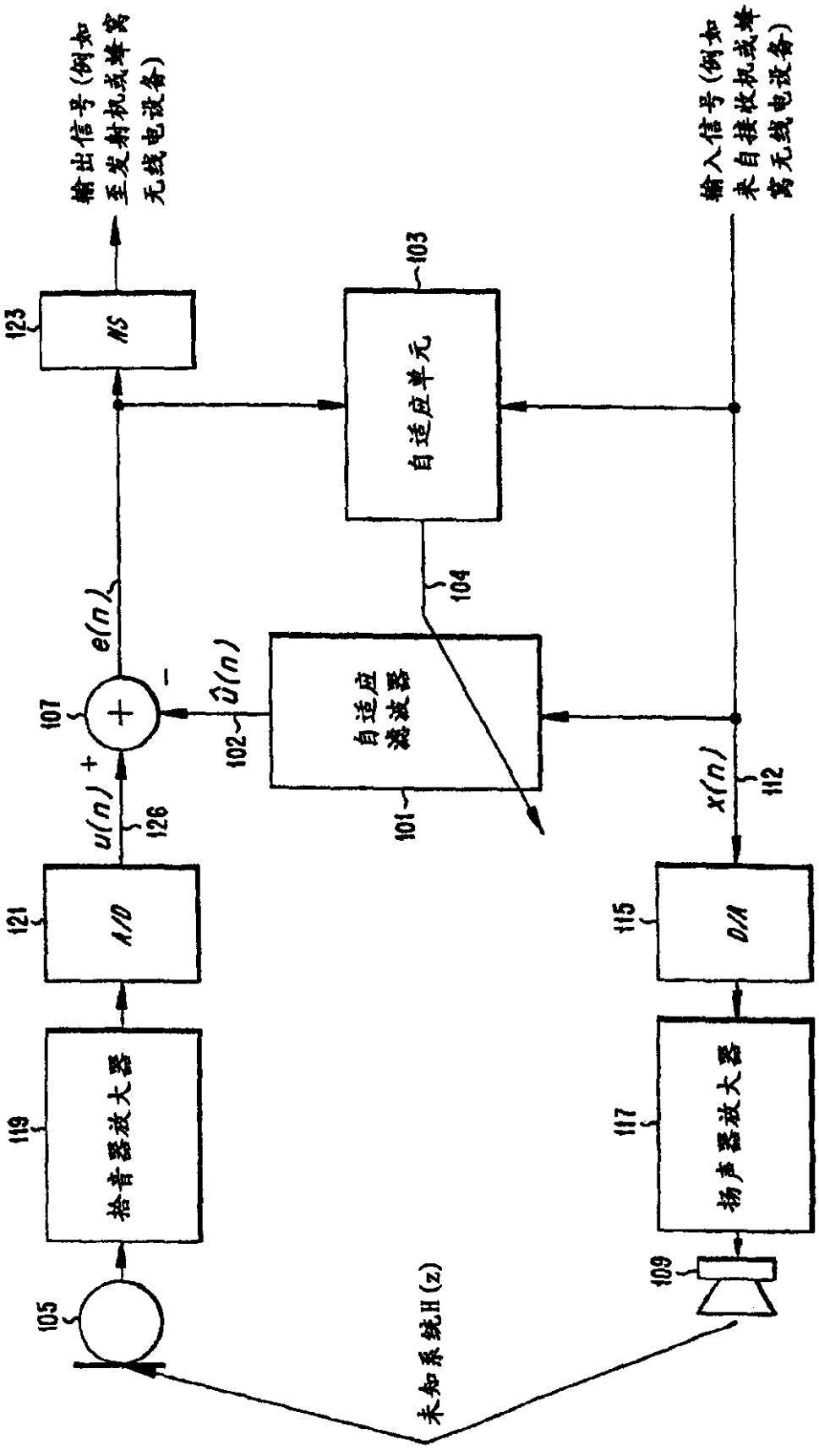



图 1
(现有技术)

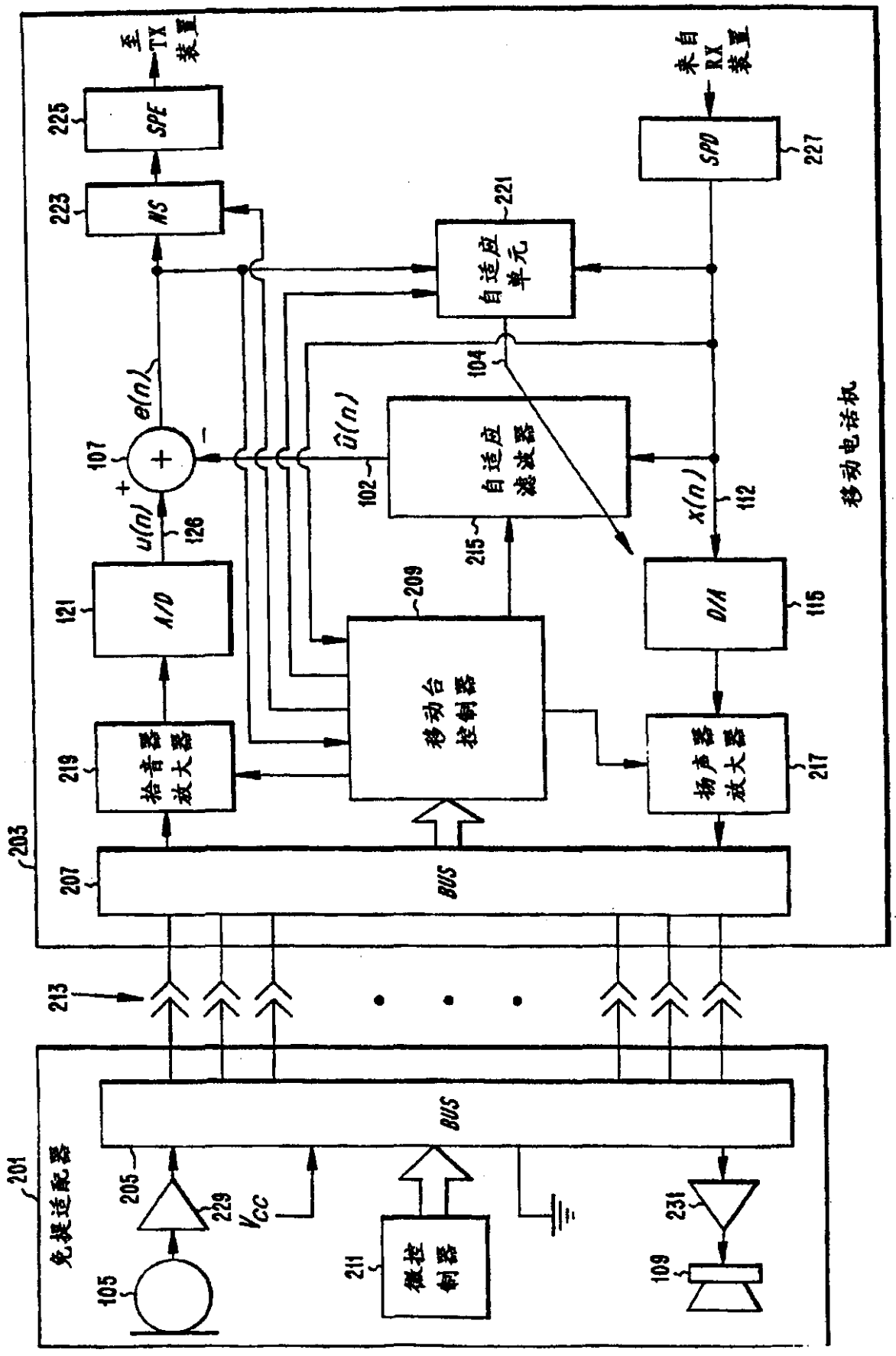

免提适配器
201
203
移动电话机
至TX装置
来自RX装置
BUS
205
207
209
211
213
215
217
219
221
223
225
227
229
231
105
109
121
126
102
104
107
112
115
$V_{CC}$
e(n)
u(n)
x(n)
SPE
MS
SPD
A/D
D/A
自适应单元
自适应滤波器
移动台控制器
拾音器放大器
扬声器放大器
微控制器

图 2

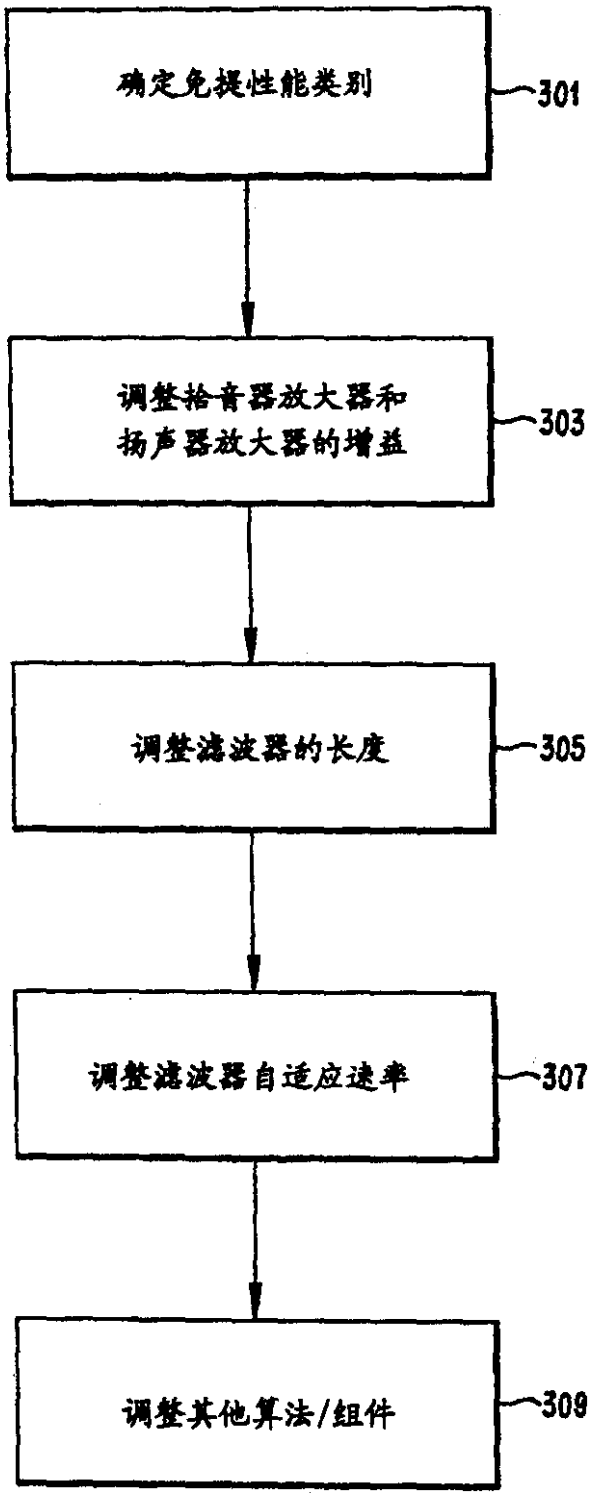

确定免提性能类别
301
调整拾音器放大器和
扬声器放大器的增益
303
调整滤波器的长度
305
调整滤波器自适应速率
307
调整其他算法/组件
309

图 3