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(54) A BACKGROUND NOISE ELIMINATE DEVICE AND METHOD FOR SPEECH COMMUNICATION TERMINAL

(57) The present invention discloses a device for eliminating background noise of a voice communication terminal, which includes a Sound Sensor Module, an Adaptive Filter, a Noise Estimating Module and an Adding Module connected successively. The invention also discloses a method for eliminating background noise of a voice communication terminal, which includes: collecting sound signals including the a talker's voice and receiver background noise received by a listener; estimating the receiver background noise signal in the signal collected; reversing the phase of the estimated receiver background noise signal and adding the estimated receiver background noise signal with reversed phase on the talker's voice signal received by the voice communication terminal before playing the added signal for the listener. The device and method can properly counteract the background noise around an earphone, and reduce the interference of background noise to the listener.

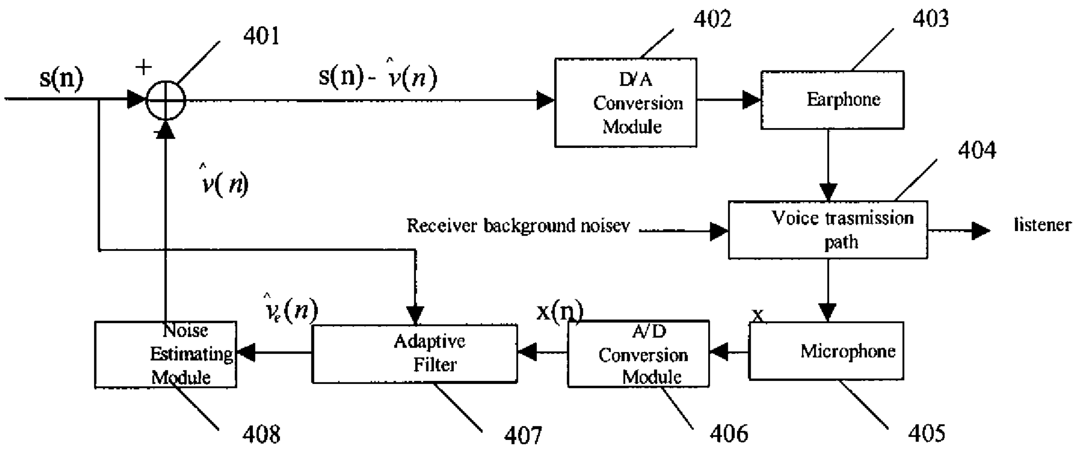

Figure 4

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Description

Field of the Technology

[0001] The present invention relates to voice processing technology of a communication terminal, and more particularly, to a device and a method for eliminating background noise of a voice communication terminal.

Background of the Invention

[0002] Along with the fast development and increasing popularization of communication technologies, communications have become an important means by which people communicate with each other; therefore, people put forward higher requirements for voice quality of calls. In a practical communication, both the talker and the listener usually communicate with each other in noisy background conditions, where the sound heard by the listener includes not only the voice of the talker, but also other sounds around the listener and the talker, such as the voice of other people, footsteps of other people, colliding sounds of objects, music and sounds emitted from vehicles. These sounds except the voice of both the talker and the listener in the communication are called background noise; the background noise around the talker is called transmitter background noise and the background noise around the listener is called receiver background noise.

[0003] Figure 1 is a schematic diagram illustrating a transmitting process of the voice of a mobile communication terminal. A microphone of the talker receives the voice of the talker and the transmitter background noise simultaneously; the received sounds, after being sampled, coded, transmitted and decoded, are sent to the earphone or speaker of the listener; the listener hears the transmitted sounds by the earphone or the speaker, and at the same time, hears the receiver background noise. Thus, it can be seen that the voice, in the transmitting process, suffers from a double interference of the transmitter background noise and the receiver background noise. In many cases, the background noise covers up the voice of the talker so that the listener can not correctly obtain the voice information sent by the talker, and thereby, resulting in difficulty in the communication.

[0004] Based on the voice transmitting process of the mobile communication terminal as shown in Figure 1, existing mobile communication terminals, such as a Code Division Multi-Address (CDMA) mobile phone or a Global System for Mobile communications (GSM) mobile phone, before a vocoder of the talker encodes the voice, an adaptive filtering technique is adopted to eliminate the transmitter background noise. Figure 2 is a schematic diagram illustrating a device for eliminating the transmitter background noise in an existing mobile communication terminal, which includes: Microphone 201, Sampling Module 202, Adaptive Filter 203 and Voice Coding Module 204.

[0005] The Microphone 201 receives the talker's voice and the background noise, and then sends the received talker's voice and background noise to the Sampling Module 202; the Sampling Module 202, upon receiving the talker's voice and the background noise, samples them, that is, converts an analogue signal into a digital signal, and outputs the obtained digital signal to the Adaptive Filter 203; the Adaptive Filter 203 will eliminate the background noise of the received signal, and output the remaining voice signal to the Voice Coding Module 204; the Voice Coding Module 204 will encode the received voice signal.

[0006] The specific structure of the Adaptive Filter 203 is shown in Figure 3, where x(n) is the sampled input signal received by the microphone, which includes a talker's voice signal s(n) and a transmitter background noise signal v(n), that is, x(n)=s(n)+v(n). After a delay Δ of x(n), x(n-Δ)= s(n-Δ) + v(n-Δ) is obtained, and x(n-Δ) passes the filter part of the Adaptive Filter 203 to output an estimation value $\hat{s}(n)$ of the voice signal s(n). Then the estimation value $\hat{s}(n)$ is subtracted from the input signal x(n) to obtain an error signal e(n), where the subtracting operation is not a general algebra subtraction, but needs to be performed according to an appropriate algorithm, such as a power spectrum density analysis of relevant power.

[0007] The Adaptive Filter 203 adopts a Least Mean Square (LMS) algorithm; and the weight value $w_i(n)$ thereof is regulated by the error signal e(n) = x(n)-$\hat{s}(n)$ = v(n) + (s(n)-$\hat{s}(n)$), $0 \le i \le M-1$, where M is the order of the filter. Supposing a weight value vector W(n) and an input vector X(n) are defined, respectively, as follows:

$$W(n) = \begin{bmatrix} w_0(n) & w_1(n) & \cdots & w_{M-2}(n) & w_{M-1}(n) \end{bmatrix}^T,$$

$$X(n) = \begin{bmatrix} x_0(n) & x_1(n) & \cdots & x_{M-2}(n) & x_{M-1}(n) \end{bmatrix}^T,$$

then the LMS algorithm of the Adaptive Filter can be expressed as:

$$\hat{s}(n) = W^T(n)X(n-\Delta) = W^T(n)S(n-\Delta) + W^T(n)V(n-\Delta),$$

$$e(n) = x(n) - \hat{s}(n),$$

$$W(n+1) = W(n) + \mu X(n)e(n),$$

where μ is a step factor for a search performed according to the LMS criterion.

[0008] Since voice has a quasi-periodicity, the voice signal s(n) and s(n-Δ) are correlated strongly; meanwhile, it is considered that the noise signal v(n) and v(n-Δ) are not correlated and that the noise signal and the voice signal are not correlated. Based on this presumption, the portion having a stronger correlation in the input signal x(n) can be estimated. At the same time, by regulating the weight value $w_i(n)$ of the filter to minimize the mean square error of e(n), an optimum estimation value $\hat{s}(n)$ of the voice signal s(n) can be obtained under the LMS criterion.

[0009] Although the solution to elimination of the transmitter background noise in the prior art can decrease the noise interference to the voice, the solution takes into account only the transmitter background noise, whereas the receiver background noise has not been dealt with, and the interference by the receiver background noise to the sound emitted from the microphone of the listener is neglected. In particular, when the listener is in a noisy environment, the receiver background noise has a severe impact on the voice of the talker. At present, the method adopted by a mobile communication terminal to overcome the receiver background noise is to increase the volume that the earphone can provide, which still has a poor effect in a noisy environment because the method is restricted by the maximal volume. Similarly, an existing fixed communication terminal has the above problems as well.

Summary of the Invention

[0010] The present invention provides a device for eliminating background noise of a voice communication terminal, which can eliminate receiver background noise so as to decrease the interference of the receiver background noise to the listener.

[0011] The present invention also provides a method for eliminating background noise of a voice communication terminal so as to reduce the interference of the receiver background noise to the listener.

[0012] A device for eliminating background noise of a voice communication terminal which includes a player used for converting input sound signal into sound and playing the sound, the device includes:

a Sound Sensor Module, used for collecting a sound played by the player as well as a receiver background noise, converting the sound played by the player and the receiver background noise into a sound signal and outputting the sound signal;

an Adaptive Filter, used for computing a noise signal comprised in the sound signal from the Sound Sensor Module, and outputting the noise signal, in which the computation is based on the input sound signal from the Sound Sensor Module and a talker's voice signal;

a Noise Estimating Module, used for estimating a receiver background noise signal based on the noise signal output by the Adaptive Filter, and outputting the estimated receiver background noise signal;

an Adding Module, used for performing an opposite-phase adding of the estimated receiver background noise signal output by the Noise Estimating Module on the voice signal from the talker, and inputting the added signal to the player of the voice communication terminal.

[0013] The sound signal output by the Sound Sensor Module is an analogue signal, and the device for eliminating background noise of a voice communication terminal further includes:

an Analogue/Digital (A/D) Converting Module, used for converting the analogue signal output by the Sound Sensor Module into a digital signal and outputting the digital signal to the Adaptive Filter.

[0014] The device for eliminating background noise of a voice communication terminal includes at least two Sound Sensor Modules.

[0015] The device for eliminating background noise of a voice communication terminal further includes an Adder, which is located between the Sound Sensor Modules and the A/D Converting Module, and is used for adding the analogue signals of multiple Sound Sensor Modules and outputting the added analogue signal to the A/D Converting Module; or located between the A/D Converting Module and the Adaptive Filter, and is used for adding the signals which have been output to the A/D Converting Module by multiple Sound Sensor Modules and have been converted by the A/D Converting Module, respectively, and outputting the added signal to the Adaptive Filter.

[0016] The A/D Converting Module of the device is used for converting the analogue signals output from multiple Sound Sensor Modules to digital signals, respectively, and outputting the digital signals to the Adaptive Filter, respectively; the Adaptive Filter is used for receiving multiple signals output from the A/D Converting Module and filtering the signals, respectively.

[0017] A method for eliminating background noise of a voice communication terminal, includes:

collecting sound signals including a talker's voice and a receiver background noise received by a listener;

estimating a receiver background noise signal from the collected signals ;

reversing the phase of the estimated receiver background noise signal;

and adding the estimated receiver background noise signal with a reversed phase on the talker's voice signal received by the voice communication terminal before playing the added signal for the listener.

[0018] The method of collecting the sound signals includes:

converting the collected sound into an analogue signal and converting the analogue signal into a digital signal via an Analogue /Digital (A/D) conversion.

[0019] The collected sound is collected through multiple paths, after the collected sound is converted into analogue signals, the method further includes the steps of:

adding the multiple analogue signals before performing the A/D conversion of the added signals;

or performing the A/D conversion of the multiple analogue signals, respectively, and adding the converted results;

or performing the A/D conversion of the multiple analogue signals, respectively, and outputting the converted results, respectively.

[0020] The step of estimating the receiver background noise signal from the collected signals includes: calculating an estimation value of the noise signal in the collected signals through the adaptive filter processing, and the adaptive filter processing using the talker's voice signal received by the voice communication terminal as the reference; and estimating a value of the receiver background noise signal using the estimation value of the noise signal.

[0021] The method of obtaining the noise signal through the adaptive filter processing includes obtaining the estimation value of the noise signal using a Least Mean Square (LMS) algorithm.

[0022] The signals collected are from multiple paths; and the step of estimating the receiver background noise signal from the collected signals includes:

performing adaptive filter processing of the multiple paths of signals, and selecting a maximum result obtained through the adaptive filter processing or the sum of all the processing results as the estimation value of the noise signal.

[0023] The method of estimating the value of the receiver background noise signal includes:

taking the sum of the estimation value of the background noise signal of the previous moment and the estimation value of the noise signal obtained through the adaptive filter processing as the estimation value of the background noise signal of the current moment.

[0024] It can be seen from the above solution that the key of the present invention lies in the Adaptive Filter for estimating the background noise and the Noise Estimation Module. The background noise is estimated through the Adaptive Filter and the Noise Estimation Module; after the phase of the background noise estimation is reversed, the background noise estimation is added on the voice signal, and then the voice signal which includes the background noise estimation is played by the earphone. As the background noise estimation and the actual receiver background noise counteract with each other, a voice signal with noise eliminated is obtained.

[0025] Therefore, the device and method for eliminating the background noise of a voice communication terminal provided by the present invention can properly counteract the background noise around the earphone, and reduce the interference of the background noise to the listener so that the listener can hear the talker's voice clearly even that the earphone volume is not high.

Brief Description of the Drawings

[0026]

Figure 1 is a schematic diagram illustrating a transmitting process of the voice of a mobile communication terminal;

Figure 2 is a schematic diagram illustrating the structure of a device for eliminating the transmitter background noise in an existing mobile communication terminal of the prior art;

Figure 3 is a schematic diagram illustrating the structure of an Adaptive Filter in the device for eliminating the transmitter background noise as shown in Figure 2;

Figure 4 is a schematic diagram illustrating the structure of a device for eliminating the receiver background noise in a mobile communication terminal according to an embodiment of the present invention;

Figure 5 is a schematic diagram illustrating the structure of the Adaptive Filter in the device for eliminating the receiver background noise as shown in Figure 4 according to an embodiment of the present invention;

Figure 6 is a schematic diagram illustrating an arrangement of an earphone and three microphones in a device for eliminating the receiver background noise according to an embodiment of the present invention;

Figure 7 is a flowchart of a method for eliminating the background noise of a voice communication terminal according to an embodiment of the present invention;

Figure 8 is a schematic diagram illustrating the structure of a device for eliminating the receiver background noise in a fixed communication terminal according to an embodiment of the present invention.

Embodiments of the Invention

[0027] The present invention is hereinafter described in detail with reference to the accompanying drawings and embodiments.

[0028] The voice communication terminal to which an embodiment of the present invention is applicable includes a player used for converting the input sound signal into the sound and playing the converted sound. The device for eliminating the background noise of the voice communication terminal, in accordance with the embodiments of the present invention, mainly includes: a Sound Sensor Module, used for collecting the sound played by the player of the voice communication terminal and the receiver background noise; an Adaptive Filter, used for computing, the noise signal contained in the sound signal from the Sound Sensor Module, according to the input sound signal from the Sound Sensor Module and the voice signal from the talker, and outputting the computed noise signal; a Noise Estimation Module, used for estimating the receiver background noise signal according to the noise signal output by the Adaptive Filter Module, and outputting the estimated receiver background noise signal; and an Adding Module, used for reversing the phase of the receiver background noise signal output by the Noise Estimation Module and adding the signal with reversed phase on the talker's voice signal before inputting the results to the player of the voice communicating terminal.

[0029] Based on the same principle, the method for eliminating the background noise of a voice communication terminal in accordance with an embodiment of the present invention mainly includes the steps as follows: collecting the sound signal received by the listener, which includes the talker's voice and the receiver background noise, estimating the receiver background noise signal from the received signal, reversing the phase of the receiver background noise signal obtained from the estimation and adding the estimated receiver background noise with reversed phase on the talker's voice signal by the voice communication terminal before playing the added sound for the listener.

[0030] The structure of the device for eliminating the receiver background noise in a mobile communication terminal according to a preferred embodiment of the present invention is shown in Figure 4, including an Adding Module 401, a Digital/Analogue (D/A) Conversion Module 402, an Earphone 403, a Microphone 405, an Analogue/Digital(A/D) Conversion Module 406, an Adaptive Filter 407 and a Noise Estimation Module 408, the Earphone 403 and the D/A Conversion Module 402 are existing components in the mobile communication terminal.

[0031] The Adding Module 401 is used for performing opposite-phase adding of the estimation value $\hat{v}(n)$ of the receiver background noise signal on the talker's voice signal s(n) received by the voice communication terminal; the D/A Conversion Module 402 is used for converting the input digital signal into an analogue signal; the Earphone 403 is used for converting the input analogue electronic signal into a sound signal and playing the sound signal, and what the Earphone 403 plays is in fact the added signal of the receiver background noise signal with reversed phase on the talker's voice signal, the earphone can be replaced by a talker; the Microphone 405 is used for receiving the talker's voice signal and the background noise which has been counteracted by the signal output by the earphone; the A/D Conversion Module 406 is used for converting the analogue signal into a digital signal; the Adaptive Filter 407 is used for estimating the noise signal value in the collected signal; and the Noise Estimation Module 408 is used for estimating the value of the background noise.

[0032] After being processed by the Adding Module 401, the talker's voice signal s(n) received by the communication terminal is transmitted to the D/A Conversion Module 402; the D/A Conversion Module 402 converts the digital signal into an analogue signal, and outputs the converted analogue signal to the Earphone 403 which is used as the player; the Earphone 403 converts the analogue signal into the sound signal and plays the converted sound signal. The sound signal played by the Earphone 403 is transmitted to the listener and the Microphone 405 used as the Sound Sensor Module through a Voice Transmission Path 404, and at the same time the receiver background noise v is transmitted to the listener and the Microphone 405 through the Voice Transmission Path 404 as well.

[0033] The Microphone 405 converts the received sound signal into an analogue signal x which includes the talker's voice signal and the receiver noise signal, and transmits the analogue signal x to the A/D Conversion Module 406; the A/D Conversion Module 406 samples the analogue signal and converts the analogue signal into a digital signal x(n) and sends digital signal x(n) to the Adaptive Filter 407; the Adaptive Filter 407 performs an Adaptive Filter processing by using the voice signal s(n) as a reference input, and estimates the value of noise signal $\hat{v}_e(n)$; the Noise Estimation Module 408 estimates the value $\hat{v}(n)$ of the background noise according to the estimation value ve(n) of the noise signal, and sends the estimation value v*(n)* of the background noise to the Adding Module 401. The A/D Conversion Module 406 is not a must here. For example, if the Adaptive Filter 407 and the Noise Estimation Module 408 can process the analogue signal, there is no need to configure the A/D Conversion Module 406; in addition, if the Microphone 405 can directly output digital signals, there is no need, either, to configure the A/D Conversion Module 406 here.

[0034] The Adding Module 401, after reversing the phase of the estimate value v(n) of the receiver background noise signal, adds the estimation value on the talker's voice signal s(n) to obtain the signal s(n) - $\hat{v}(n)$ which will be output to the D/A Conversion Module 402 and played by the Earphone 403. Thus, the estimation value -$\hat{v}(n)$ of the receiver background noise signal with reversed phase will counteract the receiver background noise v in the Voice Transmission Path 404, so that the sound sent to the listener and the Microphone 405 is mainly the talker's voice signal.

[0035] The concrete structure of the Adaptive Filter 407 is shown in Figure 5. The mixed digital signal x(n) received by the microphone, including the talker voice signal s(n) and the background noise signal, passes the filter part of the Adaptive Filter 407 and the best estimation value $\hat{s}(n)$ of the talker voice signal is recovered. The best estimation value $\hat{s}(n)$ of the voice signal minus the delay signal s(n-Δ) of the talker voice signal s(n) received by the mobile terminal through an antenna is the obtained error signal $\hat{v}_e(n)$.

[0036] The Adaptive Filter 407 adopts the LMS algorithm, of which the weight value $w_i(n)$ is regulated by the error signal $\hat{v}_e(n) - \hat{s}(n-\Delta)$, $0 \leq i \leq M-1$, where M is the order of the filter. Because of the irrelevancy of the voice signal to the background noise signal, by minimizing the mean square error of the $\hat{v}_e(n)$ through regulating the weight value $w_i(n)$, the best estimation value $\hat{s}(n)$ of the voice signal s(n) in x(n) under the criterion of Minimum Mean Square Error will be obtained; and $\hat{v}_e(n)$ is regarded as the best estimation value of the residual noise signal after the background noise is counteracted. Δ, which can be obtained through a search according to the criterion of the Minimum Mean Square Error, is the delay of the voice signal. Obviously, the Adaptive Filter 407 can adopt other adaptive algorithms and optimum criterions as well.

[0037] The Noise Estimation Module 408 estimates the value of the current background noise according to the best estimation value $\hat{v}_e(n)$ of the noise signal. The simplest estimating method is $\hat{v}(n) = v(n-1) + v_e(n)$, where $\hat{v}(n)$ is the current estimation value of the background noise, and $\hat{v}(n-1)$ is the estimation value of the background noise at the previous moment.

[0038] To make the noise counteracting effect better, multiple Earphones 403 and Microphones 405 can be configured, and moreover, the number of the earphones can be different from that of the microphones. The microphone is located where the noise is expected to be counteracted. Usually, the region where the noise is expected to be counteracted means the surrounding area of the listener's earphone. But there are exceptions in other applications, for example, when one is sleeping, if the mobile telephone is just used as a tool to reduce the surrounding noise, the region will be larger. Supposing there are one earphone and three microphones, for example, the earphone may be set in the middle, and the three microphones may be deployed with 120 degrees away from each other around the earphone as shown in Figure 6.

[0039] The signals received by multiple microphones at the same time can be transmitted to the Adaptive Filter after being added, or can be transmitted to the Adaptive Filter, respectively, and be processed, respectively; if the signals need to be added, an Adder will be correspondingly added in the device in accordance with the embodiments of the present invention. For example, signals output by the multiple microphones can be added before the A/D conversion when the Adder is placed between the microphone and the A/D Conversion Module; alternatively, they can be added after the A/D conversion when the Adder is placed between the A/D Conversion Module and the Adaptive Filter. Furthermore, the signals output by multiple microphones can be separately transmitted to the Adaptive Filter for processing, respectively, after the A/D conversion.

[0040] The embodiment of the present invention also provides a method for eliminating the receiver background noise, which is used in the listener's voice communication terminal. As shown in Figure 7, the method includes the following steps:

[0041] Step 701, collecting the voice which will be heard by the listener and which includes the receiver background noise and the voice emitted by the earphone of the voice communication terminal, converting the collected sound into the sound signal and performing the A/D conversion.

[0042] One or multiple microphones can be used for collecting the voice signal, that is, the collected signal comes from one or multiple paths. When multiple microphones are adopted to collect the signals, the output signals from multiple microphones can be added first, then the A/D conversion of the signals will be performed and the conversion results can be output; alternatively, the output signals from multiple microphones can first undergo the A/D conversion, respectively, and the conversion results will be added and then be output; or the A/D conversion of the output signals from multiple microphones can be performed, respectively, and then the results are output, respectively.

[0043] Here, the A/D conversion is not a must, for example, if the follow-on processing, such as the adaptive filter and the noise estimation, can be performed based on the analogue signals, the A/D conversion is not necessary here. Besides, if the microphone can directly output digital signals, the A/D conversion is not necessary, either.

[0044] Step 702, inputting the talker's voice signal as the reference, performing the adaptive filter processing of the collected digital voice signal, and estimating the noise signal in the collected digital voice signal, The Adaptive Filter can use the LMS algorithm or other adaptive algorithms and optimum criterions.

[0045] In the case of performing the A/D conversion of the signals output by multiple microphones respectively and then outputting the converted signals to the Adaptive Filter respectively, the adaptive filter processing needs to be performed for each path of signal, respectively. Here, supposing that there are three microphones and one earphone to give a brief description.

[0046] For example, the signals received by three microphones pass the Adaptive Filter, respectively, and are processed according to the Minimum Mean Square Error criterion. Then the estimation values of the three noise signals can be obtained, i.e., $\hat{v}_{e1}(n)$, $\hat{v}_{e2}(n)$ and $\hat{v}_{e3}(n)$, and $\hat{v}_e(n) = \max(v_{e1}(n), v_{e2}(n), v_{e3}(n))$ or $\hat{v}_e(\mathrm{n})=\mathrm{sum}(v_{e1}(n), \hat{v}_{e2}(n), \hat{v}_{e3}(n))$ can be calculated, where the max means the operation to calculate the maximum and the sum means the operation to calculate the sum. That is, the maximum of the three estimation values which are obtained after the adaptive filter processing or the sum thereof can be chosen as the output result of the Adaptive Filter. Because of the different delay of each path of signal, the estimation value of each path of noise signal, such as the $\hat{v}_{e1}(n)$, the $\hat{v}_{e2}(n)$ and the $\hat{v}_{e3}(n)$, needs to be synchronized, and then the maximum or the sum thereof is chosen.

[0047] Since the sense of human ears to voice is not in direct proportion to the voice power, but in direct proportion to the voice decibel, logarithm of the absolute value of the $\hat{v}_e(n)$ can be taken, that is, $\log|\hat{v}_e(n)|$. Then the $\log|\hat{v}_e(n)$ is minimized and the result is used as the optimum. Accordingly, in the case of multiple-path inputs, logarithm of the absolute value of each $\hat{v}_e(n)$ an be taken, and then the sum thereof will be calculated, that is, $\Sigma(\log|\hat{v}_e(n)|)$; then the $\Sigma(\log|\hat{v}_e(n)|$ is minimized and the minimized result is used as the optimum.

[0048] Step 703, estimating the value of the actual background noise signal according to the obtained noise signal, the simplest method is: $\hat{v}(n) = \hat{v}(n\text{-}1) + \hat{v}_e(n)$. The $\hat{v}(n)$ is the estimation value of the current background noise signal; the $\mathrm{v}(n\text{-}1)$ is the estimation value of the previous background noise signal; the $\hat{v}_e(n)$ is the best estimation value of the noise signal obtained through the adaptive filter processing.

[0049] Step 704, performing the phase-reversing processing for the estimation value of the actual background noise signal, and adding the estimation value of the actual background noise signal with reversed phase on the talker's voice signal before outputting the added signal via the earphone.

[0050] Step 705, the voice output by the earphone and the receiver background noise are added during the voice transmission so as to counteract the background noise therein, and the noise-eliminated talker's voice is obtained by the listener.

[0051] The solution of the embodiment of the present invention can be applied to a fixed voice communication terminal as well. The difference of the implementation lies in: it is needed to convert the received analogue signals into the digital signals before the noise estimation and elimination can be performed, because what the fixed communication terminal listener receives are the analog signals. As shown in Figure 8, an A/D Conversion Module 800 is needed before the input port of the talker's voice signal, which is used for converting the input analogue signals into the digital signals.

[0052] From the solution above, it can be found that, according to the embodiment of the present invention, residual noise after counteraction can be obtained through the adaptive filter processing, and the receiver background noise can be estimated after being continually added through the noise estimating process; then the interference of the background noise to the listener can be reduced by the counteraction of the estimated and actual receiver background noises.

[0053] To sum up, the foregoing are only preferred embodiments of the present invention and are not used for limiting the protection scope thereof. Any modification, equivalent replacement and improvement under the spirit and principle of the present invention should be covered in the protection scope of the present invention,

Claims

1. A device for eliminating background noise of a voice communication terminal which comprises a player used for converting input sound signal into sound and playing the sound, the device comprising:

   a Sound Sensor Module, used for collecting a sound played by the player as well as a receiver background noise, converting the sound played by the player and the receiver background noise into a sound signal and outputting the sound signal;
   an Adaptive Filter, used for computing a noise signal comprised in the sound signal from the Sound Sensor Module, and outputting the noise signal, in which the computation being based on the input sound signal from the Sound Sensor Module and a talker's voice signal;
   a Noise Estimating Module, used for estimating a receiver background noise signal based on the noise signal output by the Adaptive Filter, and outputting the estimated receiver background noise signal;
   an Adding Module, used for performing an opposite-phase adding of the estimated receiver background noise signal output by the Noise Estimating Module on the voice signal from the talker, and inputting the added signal to the player of the voice communication terminal.

2. The device according to Claim 1, wherein the sound signal output by the Sound Sensor Module is an analogue signal, the device further comprising:

   an Analogue/Digital (A/D) Converting Module, used for converting the analogue signal output by the Sound Sensor Module into a digital signal and outputting the digital signal to the Adaptive Filter.

3. The device according to Claim 2, wherein the device comprises at least two Sound Sensor Modules.

4. The device according to Claim 3, the device further comprising an Adder which is located between the Sound Sensor Modules and the A/D Converting Module, and is used for adding the analogue signals of multiple Sound Sensor Modules and outputting the added analogue signal to the A/D Converting Module;
   or located between the A/D Converting Module and the Adaptive Filter, and is used for adding the signals which have been output to the A/D Converting Module by multiple Sound Sensor Modules and have been converted by the A/D Converting Module, respectively, and outputting the added signal to the Adaptive Filter.

5. The device according to Claim 3, wherein the A/D Converting Module of the device is used for converting the analogue signals output from multiple Sound Sensor Modules to digital signals, respectively, and outputting the digital signals to the Adaptive Filter, respectively; the Adaptive Filter is used for receiving multiple signals output from the A/D Converting Module and filtering the signals, respectively.

6. A method for eliminating background noise of a voice communication terminal, comprising:

   collecting sound signals comprising a talker's voice and a receiver background noise received by a listener;
   estimating a receiver background noise signal from the collected signals ;
   reversing the phase of the estimated receiver background noise signal;
   and adding the estimated receiver background noise signal with a reversed phase on the talker's voice signal received by the voice communication terminal before playing the added signal for the listener.

7. The method according to Claim 6, wherein the method of collecting the sound signals comprises:

converting the collected sound into an analogue signal and converting the analogue signal into a digital signal via an Analogue /Digital (A/D) conversion.

8. The method according to Claim 7, wherein the collected sound is collected through multiple paths, after the collected sound is converted into analogue signals, further comprising:

adding the multiple analogue signals before performing the A/D conversion of the added signals;
or performing the A/D conversion of the multiple analogue signals, respectively, and adding the converted results;
or performing the A/D conversion of the multiple analogue signals, respectively, and outputting the converted results, respectively.

9. The method according to any one from Claim 6 to Claim 8, wherein the step of estimating the receiver background noise signal from the collected signals comprises: calculating an estimation value of the noise signal in the collected signals through the adaptive filter processing, and the adaptive filter processing using the talker's voice signal received by the voice communication terminal as the reference; and estimating a value of the receiver background noise signal using the estimation value of the noise signal.

10. The method according to Claim 9, wherein the method of obtaining the noise signal through the adaptive filter processing comprises obtaining the estimation value of the noise signal using a Least Mean Square (LMS) algorithm.

11. The method according to Claim 9, wherein the signals collected are from multiple paths; and the step of estimating the receiver background noise signal from the collected signals comprises:

performing adaptive filter processing of the multiple paths of signals, and selecting a maximum result obtained through the adaptive filter processing or the sum of all the processing results as the estimation value of the noise signal.

12. The method according to Claim 9, wherein the method of estimating the value of the receiver background noise signal comprises:

taking the sum of the estimation value of the background noise signal of the previous moment and the estimation value of the noise signal obtained through the adaptive filter processing as the estimation value of the background noise signal of the current moment.

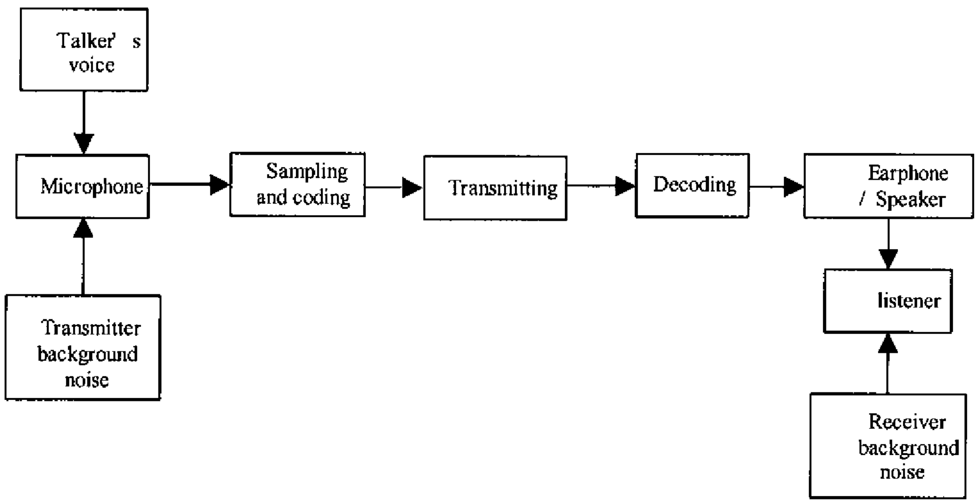

Figure 1

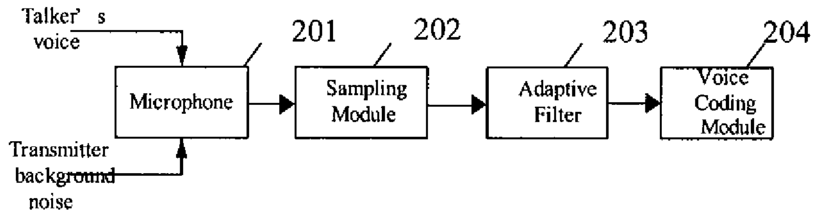

Figure 2

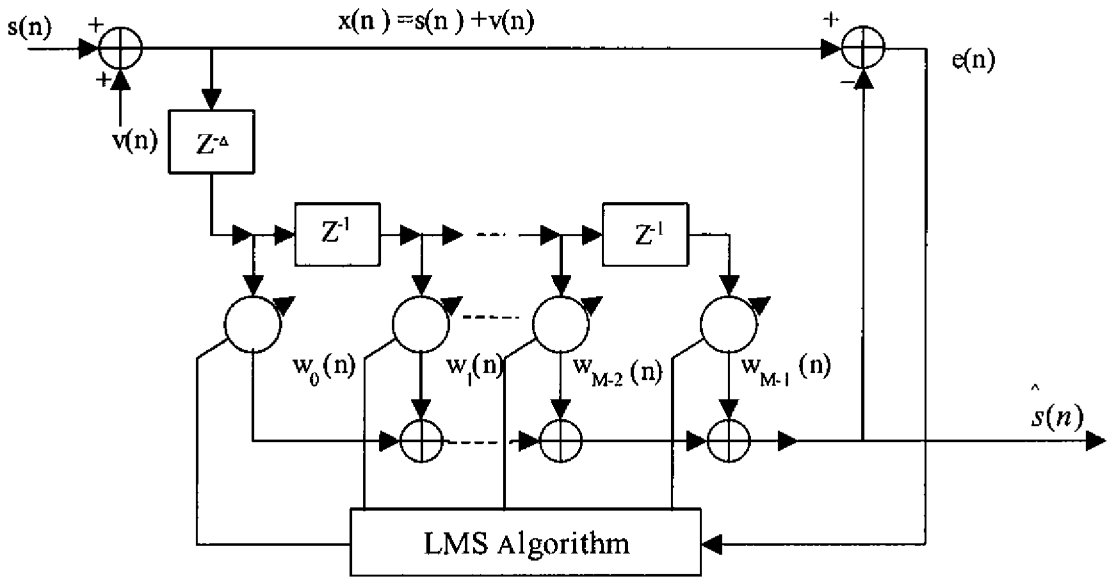

Figure 3

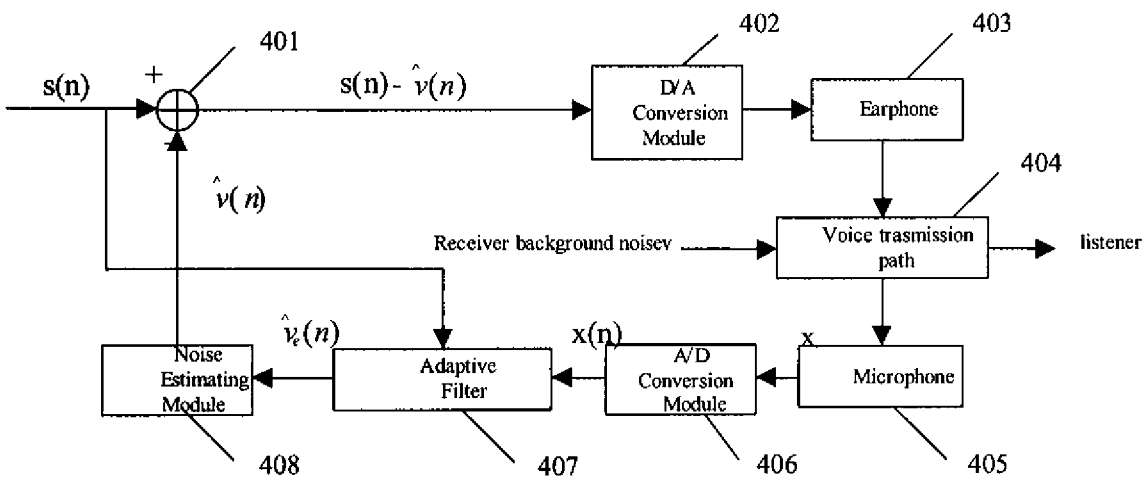

Figure 4

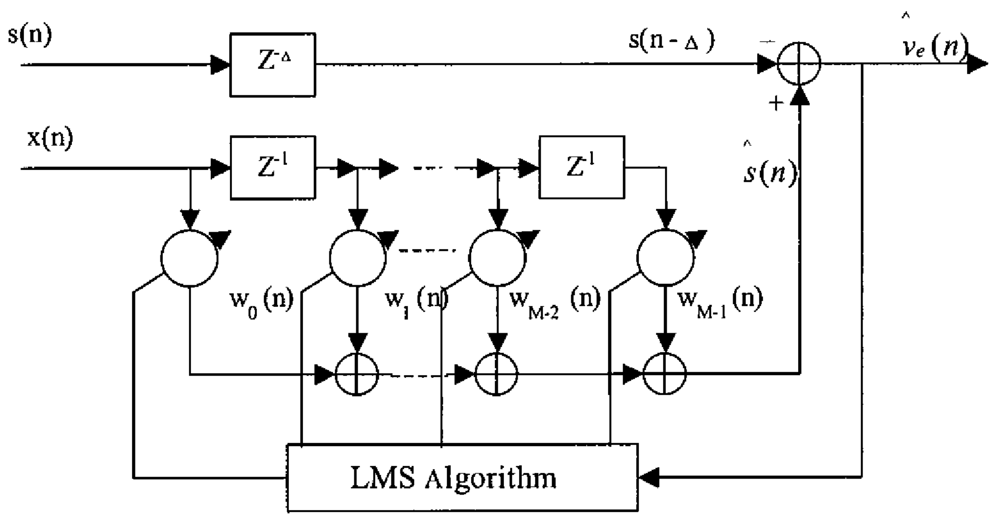

Figure 5

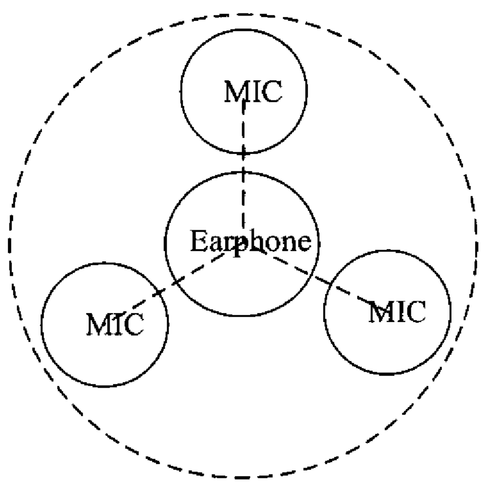

Figure 6

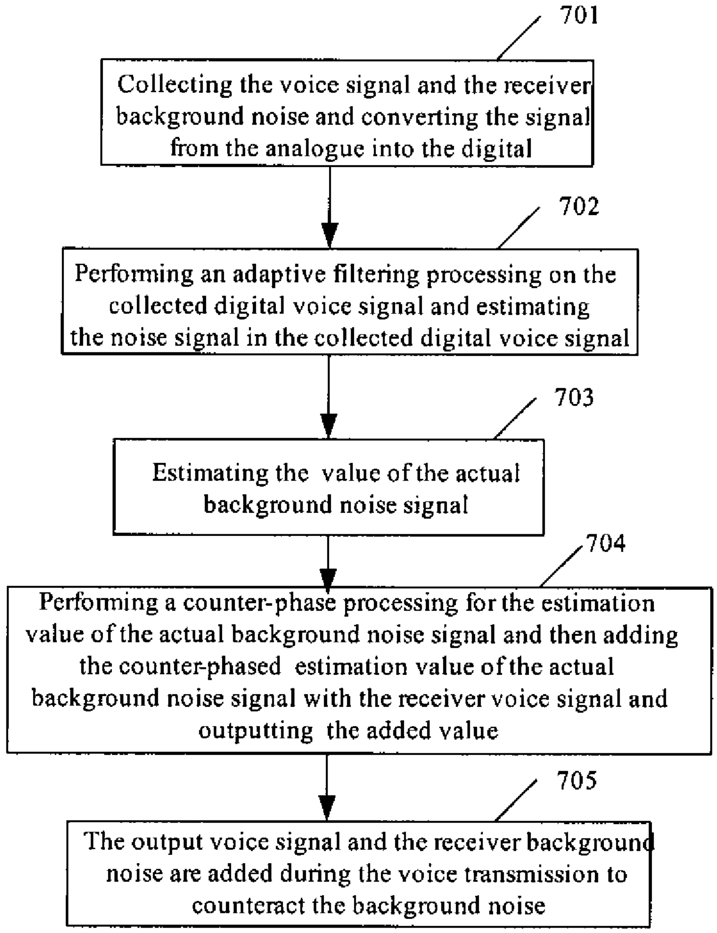

Figure 7

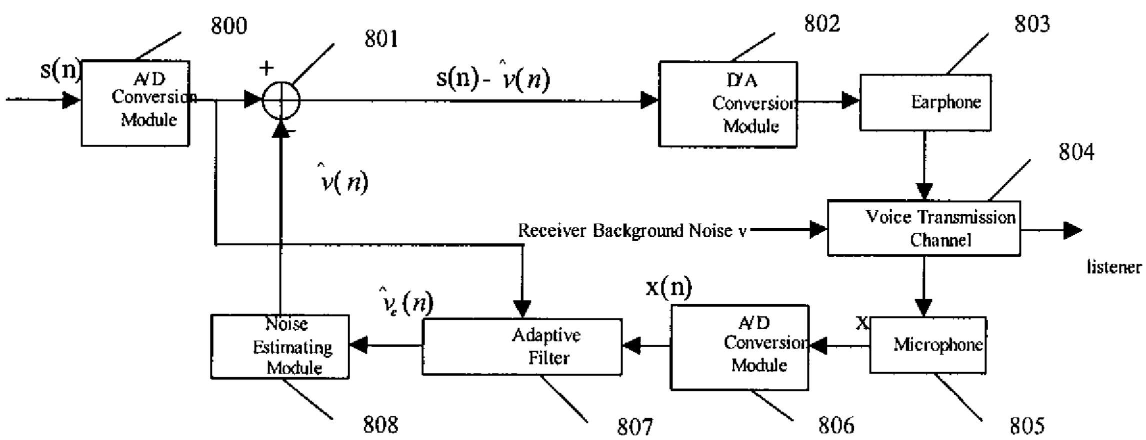

Figure 8

INTERNATIONAL SEARCH REPORT

International application No.
PCT/CN2005/001289

A. CLASSIFICATION OF SUBJECT MATTER

IPC[7] G10L 21/02 H04B 1/10

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC[7] G10L 21/+ H04B 1/+ H04M 1/+

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

The Chinese patent document (1985-)

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

CNPAT CNKI WPI EPODOC PAJ background noise cancel eliminate suppress speech communication terminal adaptive filter estimate counteract adder

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	CN,Y,2256610 (HUAYUE COMMUNICATION EQUIPMENT LTD, SHAOXING) 18.JUN.1997 (18.06.1997) see whole document	1-3, 6-7
Y	CN,A,1152830 (MATSUSHITA ELECTRIC IND CO LTD) 25.JUN.1997 (25.06.1997) see line 8, page 5- line 11, page 6 ,figure 1	1-3, 6-7
A	CN,A,1415139 (TELEFONAKTIEBOLAGET ERICSSON L M) 30.APR.2003 (30.04.2003) see whole document	1-12

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

* Special categories of cited documents:

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

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

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

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

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

"&"document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
12.OCT.2005 (12.10.2005)	01 · DEC 2005 (0 1 · 1 2 · 2 0 0 5)

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INTERNATIONAL SEARCH REPORT
Information on patent family members

International application No.
PCT/CN2005/001289

Patent Documents referred in the Report	Publication Date	Patent Family	Publication Date
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		JP,A,9036784	07.FEB.1997(07.02.1997)
CN,A,1415139	30.APR.2003 (30.04.2003)	US,B1,6865276	08.MAR.2005(08.03.2005)
		WO,A2,137439	25.MAY.2001(25.05.2001)
		AU,A,200113883	30.MAY.2001(30.05.2001)
		EP,A2,1228572	07.AUG.2002(07.08.2002)
		JP,T,2003514493	15.APR.2003(15.04.2003)

Form PCT/ISA /210 (patent family annex) (April 2005)