

A Control System and Control Method for Hearing Detection

Technical field

[0001] The present invention relates to the technical field of medical devices, and particularly a control system and control method for hearing detection.

Background technique

[0002] With the increase of age, people's hearing declines, especially for the elderly, whose lives have been seriously affected by their hearing ability. One of the hearing aids has a device to detect if there is a telephone receiver near the hearing aid. Specifically, a sensitive element is provided to detect the voice coil of the telephone receiver when the receiver is kept close to the ear of the user who wears the corresponding hearing aid. When both hearing aids operate in a normal hearing mode, this constitutes a microphone mode, that is, the user amplifies any noise or sound around him, which seriously affects the sound quality.

[0003] In summary, the problem with the prior art is that when the existing hearing aid operates in a normal hearing mode, it will constitute a microphone mode, that is, the user amplifies any noise or sound around it, which seriously affects the sound quality. Moreover, the accuracy of hearing data obtained by the prior art is poor.

Summary of the invention

[0004] In view of the problems with the prior art, the present invention provides a control system and control method for hearing detection.

[0005] The present invention is achieved through a hearing detection control system which includes:

[0006] An acquisition unit for detecting surrounding sound information;

[0007] The acquisition unit receives the returned audio signal by the microphone, and uses a band-pass filter to filter out unconcerned frequency bands. Use the window function to reduce the influence of spectrum leakage, take fast Fourier transform to obtain the frequency domain signal, and process it as decibel data and store it in the sampling queue Queue1;

[0008] Obtain the spectrum analysis window queue w1 from the sampling queue. The calculation formula is:

$$[0009] \quad w_1(x) = Queue_1(x + [(f_0 - f_{shift}) * f_{min}]), x \in [0, 2 * [f_{shift} * f_{min}]] ;$$

[0010] Of which, Queue₁ is the sampling queue;

[0011] f_0 is the center frequency of the audio signal;

[0012] f_{shift} is the maximum value of frequency change;

[0013] f_{min} is the minimum resolution of the spectral signal;

[0014] Obtain the left mean value W_{left} and the right mean value W_{right} of the maximum frequency shift f_{shift} from the spectrum analysis window queue:

$$[0015] \quad W_{left} = \frac{\sum_{i=0}^{[f_{shift} * f_{min}] - 2} w_1(i)}{[f_{shift} * f_{min}] - 2} ;$$

$$[0016] \quad W_{right} = \frac{\sum_{i=[f_{shift} * f_{min}] + 2}^{2 * [f_{shift} * f_{min}]} w_1(i)}{[f_{shift} * f_{min}] - 2} ;$$

[0017] The cycle $T_{simulate}$ of the simulation definition logic. If $T_{simulate}$ is too long, the user will have an obvious sense of delay, and the human-computer interaction will be very poor. However, if $T_{simulate}$ is too short, the sampling data cannot be updated in time, there will be a lot of wrong

operations. According to the concept of definition, the inevitable error rate R_{error} is: due to the difference between the signal sampling frequency and the acquisition signal processing and analysis frequency, an inevitable error will occur. The formula of T_{simulate} is:

$$[0018] \quad T_{\text{simulate}} = \frac{T_0}{1 + R_{\text{error}}} \quad ;$$

[0019] Of which, T_0 is the sampling period;

[0020] R_{error} is the inevitable error rate;

[0021] Define a threshold T_{Degree} that is affected by the energy E_{peak} of the center frequency f_0 and the weight α , calculate the change rate of the left mean value w_{left} and the right mean value w_{right} in a unit T_{simulate} period, and compare it with the threshold T_{Degree} :

$$[0022] \quad T_{\text{Degree}} = \frac{\alpha E_{\text{peak}}}{[f_{\text{shift}} * f_{\text{min}}]} \quad \alpha \in [0, 2] \quad ;$$

[0023] Of which, α is the weight;

[0024] E_{peak} is the energy of the center frequency f_0 ;

$$[0025] \quad R_{\text{left}} = \frac{\Delta w_{\text{left}}}{T_0} \quad ;$$

$$[0026] \quad R_{\text{right}} = \frac{\Delta w_{\text{right}}}{T_0} \quad ;$$

[0027] Comparison process:

[0028] $R_{\text{left}} > T_{\text{Degree}}$;

[0029] What happens is an action of moving away from the microphone, the message is recorded as L_{event} to the message queue $\text{Queue}_{\text{event}}$:

[0030] $R_{\text{right}} > T_{\text{Degree}}$;

[0031] What happens is an action of moving close to the microphone, the message is recorded as R_{event} to the message queue $\text{Queue}_{\text{event}}$:

[0032] $R_{\text{left}} > T_{\text{Degree}}$ and $R_{\text{right}} > T_{\text{Degree}}$;

[0033] What happens is a compound action (for example, a flip gesture, or back and forth movement of both hands), and the message is recorded as D_{event} to the message queue $\text{Queue}_{\text{event}}$;

[0034] Within the waiting time of T_{delay} , if a message is generated, the message is recorded to the message queue $\text{Queue}_{\text{event}}$, the timer is reset and wait for the T_{delay} time again; if no message is generated, a message processing analysis will be performed on the message queue $\text{Queue}_{\text{event}}$, and the queue will be cleared;

[0035] If there is no timer resetting within a certain T_{delay} time, there will be a message processing analysis on the message queue $\text{Queue}_{\text{event}}$ to obtain the message logic. Simulate the operation reflected by the message logic to achieve the control of the computer and complete the human-computer interaction;

[0036] The conductor is electrically connect to the acquisition unit to transmit sound information;

[0037] The filtering device is connect to the conductor to filter sound signals;

[0038] The filtering device performs down-conversion processing on the received intermediate frequency signal $r(n)$:

$$\begin{aligned}
 [0039] \quad s_r(n) &= r(n) * e^{-j2\pi i * \Delta f} \\
 &= \{h(l) * c(n) * e^{j*2\pi f_1} + P(n)\} * e^{-j2\pi i * \Delta f} ;
 \end{aligned}$$

[0040] Among them, $s_r(n)$ is the baseband signal, $P(n)$ is the noise of the intermediate frequency signal $r(n)$ and the noise is complex Gaussian noise. The real and virtual parts obey the Gaussian distribution with mean value of 0 and the variance of δ^2 , f_1 is the signal frequency, Δf is the search step length, i is the number of searches. Divide the $s_r(n)$ signal into Q ms data represented as $s_{r1}(n), \dots, s_{rq}(n), \dots, s_{rQ}(n)$, and multiply each of them respectively with the complex conjugate of the local code frequency domain value, and then take the inverse Fourier transform, which is represented as:

$$[0041] \quad \begin{cases} R_1(n) = IFFT[FFT(s_{r1}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_q(n) = IFFT[FFT(s_{rq}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_Q(n) = IFFT[FFT(s_{rQ}(n)) * conj(FFT(c(n)))] \end{cases};$$

[0042] Among them, $FFT()$ represents fast Fourier transform of the received signals, $IFFT()$ represents the fast Fourier transform of the acquired signals, $conj()$ represents the complex conjugate of the acquired signal, and $c(n)$ represents the local C/A code;

[0043] An amplifier that is electrically connected to the filter device for amplifying the sound signal;

[0044] The transfer function of the amplifier provided with the sub-matched filter is:

$$H(z) = \sum_{i=0}^{N-1} h_i z^{-i} = \sum_{i=0}^{N-1} x_{N-1-i} Z^{-i} = X(z^{-1}) Z^{-(N-1)}$$

C_i is modulated by the hierarchical sequence u and v , u is the hierarchical Golay sequence $u = \{1, 1, 1, 1, 1, 1, 1, -1, -1, 1, -1, 1, -1, 1, -1, -1, 1\}$, $v = \{1, 1, 1, -1, -1, 1, -1, -1, 1, 1, 1, -1, 1, -1, 1, 1\}$, $C_{16m+n} = u_n v_m$;

$$[0045] \quad H(z) = X(z) = C(z) = \sum_{i=0}^{L_u L_v - 1} C_i z^{-i} = \sum_{i=0}^{L_u L_v - 1} C_{16m+n} z^{-(16m+n)} = \sum_{n=0}^{L_u-1} u_n z^{-n} \sum_{m=0}^{L_v-1} v_m z^{-16m} = H(z_u) H(z_v), \quad \text{According}$$

to the hierarchical Golay sequence to improve the transfer function, there comes:

[0046] $H(z_u)=[1+z^{-8}+z^{-1}(1-z^{-8})][1+z^{-4}+z^{-2}(1-z^{-4})];$

[0047] $H(z_v)=(1+z^{-1})[1-z^{-6}+z^{-8}+z^{-14}]+(1-z^{-1})[z^{-2}-z^{-4}+z^{-10}+z^{-12}];$

[0048] An analyzer is electrically connected to the amplifier to compare the amplified signal with the corresponding computer data signal, and analyze the sound generated by the vibration frequency and the vibration signal of the current user at the current time;

[0049] The data compression method of the analyzer contains the following steps:

[0050] Step 1. When coding, first calculate the E1 value according to the formula

$$\Delta d'_n = \frac{4 \times d_n}{n} - \frac{6 \times E1_n}{n(n+1)}$$

$E1_{n+1} = E1_n + d_{n+1}$, and then calculate the fitting residual according to

and $DFR_n = \Delta d'_n - \lfloor \Delta d'_n \rfloor = \varepsilon$. When calculating these two, you need to make an overrun judgment on the result to determine whether E1 is over-limit, so as to avoid overflow caused by exceeding the upper limit of the sensor data bus. Determining whether the residual is over-limit is to achieve piecewise fitting;

[0051] Step 2. After all the fitting residuals of a piece of input data are calculated, there will be data packet generated in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$. Make entropy coding by the S-Huffman coding method, and send it out. For the decoding at the receiving end, a set of data for decoding will be received and the data packet shown in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$, will be restored. Calculate and restore all the original data according to the formula

$$d_{n-1} = d_n - \lfloor \Delta d'_n \rfloor - DFR_n = d_n - \left\lfloor \frac{4 \times d_n}{n} - \frac{6 \times E1_n}{n(n+1)} \right\rfloor - DFR_n$$

[0052] An input converter is electrically connected to the analyzer and used to convert an input signal into an electric input signal;

[0053] An output converter is electrically connected with the converter, and is used to convert the electric output signal into the audio output signal.

[0054] Furthermore, an amplifier which includes:

[0055] A digitization module for digitizing the input signal;

[0056] The nonlinear compensation database module is connected with a digitization module, a output signal converter module, and a nonlinear compensation database module, and used to perform nonlinear digitization compensation processing on the input signal through the digitization module and the feedback signal through the nonlinear compensation database module;

[0057] The output signal converter module is connected with the nonlinear digital compensation module and the output circuit, and used to convert the input signal through the nonlinear digital compensation module and the signal fed back by the upper and lower electromagnetic coils into a signal suitable for system control;

[0058] The output circuit module connected with the output signal converter and the upper and lower electromagnetic coils to output the signals processed by the signal converter;

[0059] The position signal module is connected with the position sensor and the input interface module and used to process the position signal detected by the position sensor and transmit it to the input interface module;

[0060] The input interface is connected with the position signal module and the compensation parameter determination module and used to transmit the signal of the position signal module to the input interface of the compensation parameter determination module;

[0061] The compensation parameter determination module is connected with the input interface module, material parameter input module, and nonlinear compensation database module and used

to comprehensively process the signal transmitted by the upper and lower coils and the position feedback signal transmitted by the input interface module according to the reference data provided by the material parameter input module;

[0062] The material parameter input module is connected with the compensation parameter determination module and used to provide and input the material parameter input module for the compensation parameter determination module;

[0063] Nonlinear compensation database module is connected with the nonlinear digitization compensation module and compensation parameter determination module and used to process the feedback signal transmitted by the compensation parameter determination module and input it into the nonlinear digital compensation module.

[0064] A control method for a hearing detection control system as described in claim 1, is specially based on a network architecture detection control module built in the acquisition unit to perform control, which specifically includes:

[0065] There are decision indicators $p_1, p_2, \dots, p_m$ in the listening target and decision-making layer, and the network architecture layer under the listening target and decision-making layer has $C_1, C_2, \dots, C_N$ index sets, among which C_i contains the element $e_{i1}, e_{i2}, \dots, e_{in_i}, i = 1, 2, \dots, N$.

[0066] Furthermore, the control method is also characterized by taking the hearing target and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , that is, to construct a judgment matrix under the criterion p_s :

[0067]	e_{jk}	$e_{i1}, e_{i2}, \dots, e_{in_i}$	归一化特征向量	Normalized Feature Vector
	e_{i1}		$w_{i1}^{(jk)}$	
	e_{i2}		$w_{i2}^{(jk)}$	
	$\vdots$		$\vdots$	
	e_{in_i}		$w_{in_i}^{(jk)}$	

[0068] And get the weight vector by the characteristic root method $w_{i1}^{(jk)}, w_{i2}^{(jk)}, \dots, w_{in_i}^{(jk)}$ 。

[0069] Furthermore, the control method is also featured by

[0070] Repeating the above steps for $k=1, 2, \dots, n_i$ to obtain the matrix W_{ij} , which is shown below.

$$[0071] \quad W_{ij} = \begin{pmatrix} w_{i1}^{(j1)} & w_{i1}^{(j2)} & \dots & w_{i1}^{(jn_j)} \\ w_{i2}^{(j1)} & w_{i2}^{(j2)} & \dots & w_{i2}^{(jn_j)} \\ \vdots & \vdots & \vdots & \vdots \\ w_{in_i}^{(j1)} & w_{in_i}^{(j2)} & \dots & w_{in_i}^{(jn_j)} \end{pmatrix},$$

[0072] Of which, the column vector of W_{ij} is the ranking vector of the influence degree of the elements $e_{i1}, e_{i2}, \dots, e_{in_i}$ in C_i on the elements $e_{j1}, e_{j2}, \dots, e_{jn_j}$ in C_j . If the elements in C_j are not affected by the elements in C_i , $W_{ij}=0$.

[0073] The control method of the hearing detection control system as described in claim 3 is also characterized by repeatedly taking the listening target for $i=1, 2, \dots, N$; $j=1, 2, \dots, N$ and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , to obtain the super matrix W under the decision criterion p_s :

$$[0074] \quad W = \begin{pmatrix} C_1 \\ C_2 \\ \vdots \\ C_N \end{pmatrix} \begin{pmatrix} W_{11} & W_{12} & \cdots & W_{1N} \\ W_{21} & W_{22} & \cdots & W_{2N} \\ \vdots & \vdots & \ddots & \vdots \\ W_{N1} & W_{N2} & \cdots & W_{NN} \end{pmatrix}$$

[0075] The control method of the hearing detection control system as described in claim 3 is also characterized by reflecting one-step dominance of the element i over the element j in the super matrix W . Calculating W^2 . The element W_{ij}^2 represents the two-step dominance of element i over element j . W^2 is still listed as a normalized matrix. Calculating W^3 , W^4 , ..., in the same manner. When W^∞ exists, the j th column of W^∞ is the limit relative weight vector of each element to j in the network architecture layer under the criterion p_s ,

$$[0076] \quad W^\infty = \begin{pmatrix} W_{11}^\infty & W_{12}^\infty & \cdots & W_{1N}^\infty \\ W_{21}^\infty & W_{22}^\infty & \cdots & W_{2N}^\infty \\ \vdots & \vdots & \ddots & \vdots \\ W_{N1}^\infty & W_{N2}^\infty & \cdots & W_{NN}^\infty \end{pmatrix}$$

[0077] Of which, the value of each row is the local weight vector of the corresponding element; when a row is all 0, the corresponding local weight is 1; arrange the local weights in the order of elements to obtain the local weight vector;

$$[0078] \quad Q = [q_{11}, \cdots, q_{1n_1}, q_{21}, \cdots, q_{2n_2}, \cdots, q_{N1}, \cdots, q_{Nn_N}]^T \circ$$

[0079] The control method of the hearing detection control system as described in claim 3 is characterized by the decision index which comprises the extremely small hearing value, the very small hearing value, the small hearing value, the relatively large hearing value, and the large hearing value.

[0080] The advantages and positive effects of the present invention are: the sound signal is detected by the acquisition unit, and conducted through the penetration. The impurity sound is filtered out by the filter device. The normal sound signal is amplified by the amplifier, and analyzed and further screened and processed by the analyzer, and finally transmitted through the input converter and output converter to provide users with increased comfort.

[0081] The control method of the present invention can obtain more accurate hearing data.

Description of the drawings

[0082] FIG. 1 is a schematic structural diagram of a control system of a lighting device based on an embodiment of the present invention;

[0083] In the figure: 1. Acquisition unit; 2. Conductor; 3. Filter device; 4. Amplifier; 5. Analyzer; 6. Input converter; 7. Output converter.

Detailed description

[0084] In order to make the objectives, technical solutions, and advantages of the present invention clearer, the present invention will be further described in detail below in conjunction with embodiments. It should be understood that the specific embodiments described here are only used to explain the present invention, but not to limit the present invention.

[0085] As shown in FIG. 1, the hearing detection control system based on the embodiment of the present invention includes:

[0086] An acquisition unit 1 for detecting surrounding sound information;

[0087] A conductor 2 is electrically connected to the acquisition unit 1 for transmitting sound information;

[0088] A filter device 3 is electrically connected to the conductor 2 for filtering sound signals;

[0089] An amplifier 4 is electrically connected to the filtering device 3 for amplifying sound signals;

[0090] An analyzer 5 is electrically connected to the amplifier 4 and used to compare the amplified signal with the corresponding computer data signal, analyze the sound generated by the vibration frequency and the vibration signal of the current user at the current time;

[0091] An input converter 6 is electrically connected to the analyzer 5 and used to convert an input signal into an electrical input signal;

[0092] An output converter 7 is electrically connected to the input converter 6 and used to convert an electric output signal into an audio output signal.

[0093] The acquisition unit 1 receives the returned audio signal by the microphone, and uses a band-pass filter to filter out unconcerned frequency bands. Use the window function to reduce the influence of spectrum leakage, take fast Fourier transform to obtain the frequency domain signal, and process it as decibel data and store it in the sampling queue Queue1;

[0094] Obtain the spectrum analysis window queue w1 from the sampling queue. The calculation formula is:

$$[0095] \quad w_1(x) = Queue_1(x + [(f_0 - f_{shift}) * f_{min}]), x \in [0, 2 * [f_{shift} * f_{min}]] ;$$

[0096] Of which, Queue₁ is the sampling queue;

[0097] f_0 is the center frequency of the audio signal;

[0098] f_{shift} is the maximum value of frequency change;

[0099] $f_{\min}$ is the minimum resolution of the spectral signal;

[0100] Obtain the left mean value W_{left} and the right mean value W_{right} of the maximum frequency shift f_{shift} from the spectrum analysis window queue:

$$[0101] \quad W_{\text{left}} = \frac{\sum_{i=0}^{\lfloor f_{\text{shift}} * f_{\min} \rfloor - 2} w_1(i)}{\lfloor f_{\text{shift}} * f_{\min} \rfloor - 2} ;$$

$$[0102] \quad W_{\text{right}} = \frac{\sum_{i=\lfloor f_{\text{shift}} * f_{\min} \rfloor + 2}^{2 * \lfloor f_{\text{shift}} * f_{\min} \rfloor} w_1(i)}{\lfloor f_{\text{shift}} * f_{\min} \rfloor - 2} ;$$

[0103] The cycle T_{simulate} of the simulation definition logic. If T_{simulate} is too long, the user will have an obvious sense of delay, and the human-computer interaction will be very poor. However, if T_{simulate} is too short, the sampling data cannot be updated in time, there will be a lot of wrong operations. According to the concept of definition, the inevitable error rate R_{error} is: due to the difference between the signal sampling frequency and the acquisition signal processing and analysis frequency, an inevitable error will occur. The formula of T_{simulate} is:

$$[0104] \quad T_{\text{simulate}} = \frac{T_0}{1 + R_{\text{error}}} ;$$

[0105] Of which, T_0 is the sampling period;

[0106] R_{error} is the inevitable error rate;

[0107] Define a threshold T_{Degree} that is affected by the energy E_{peak} of the center frequency f_0 and the weight α , calculate the change rate of the left mean value w_{left} and the right mean value w_{right} in a unit T_{simulate} period, and compare it with the threshold T_{Degree} :

$$[0108] \quad T_{\text{Degree}} = \frac{\alpha E_{\text{peak}}}{\lfloor f_{\text{shift}} * f_{\min} \rfloor} \quad \alpha \in [0, 2] ;$$

[0109] Of which, α is the weight;

[0110] E_{peak} is the energy of the center frequency f_0 ;

$$[0111] R_{\text{left}} = \frac{\Delta w_{\text{left}}}{T_0} ;$$

$$[0112] R_{\text{right}} = \frac{\Delta w_{\text{right}}}{T_0} ;$$

[0113] Comparison process:

[0114] $R_{\text{left}} > T_{\text{Degree}}$;

[0115] What happens is an action of moving away from the microphone, the message is recorded as L_{event} to the message queue $\text{Queue}_{\text{event}}$:

[0116] $R_{\text{right}} > T_{\text{Degree}}$;

[0117] What happens is an action of moving close to the microphone, the message is recorded as R_{event} to the message queue $\text{Queue}_{\text{event}}$:

[0118] $R_{\text{left}} > T_{\text{Degree}}$ and $R_{\text{right}} > T_{\text{Degree}}$;

[0119] What happens is a compound action (for example, a flip gesture, or back and forth movement of both hands), and the message is recorded as D_{event} to the message queue $\text{Queue}_{\text{event}}$;

[0120] Within the waiting time of T_{delay} , if a message is generated, the message is recorded to the message queue $\text{Queue}_{\text{event}}$, the timer is reset and wait for the T_{delay} time again; if no message is generated, a message processing analysis will be performed on the message queue $\text{Queue}_{\text{event}}$, and the queue will be cleared;

[0121] If there is no timer resetting within a certain T_{delay} time, there will be a message processing analysis on the message queue $\text{Queue}_{\text{event}}$ to obtain the message logic. Simulate the operation reflected by the message logic to achieve the control of the computer and complete the human-computer interaction;

[0122] The filtering device 3 performs down-conversion processing on the received intermediate frequency signal $r(n)$:

$$[0123] \quad s_r(n) = r(n) * e^{-j2\pi i \Delta f} \\ = \{h(l) * c(n) * e^{j2\pi f_1} + P(n)\} * e^{-j2\pi i \Delta f};$$

[0124] Among them, $s_r(n)$ is the baseband signal, $P(n)$ is the noise of the intermediate frequency signal $r(n)$ and the noise is complex Gaussian noise. The real and virtual parts obey the Gaussian distribution with mean value of 0 and the variance of δ^2 , f_1 is the signal frequency, Δf is the search step length, i is the number of searches. Divide the $s_r(n)$ signal into Q ms data represented as $s_{r1}(n), \dots, s_{rq}(n), \dots, s_{rQ}(n)$, and multiply each of them respectively with the complex conjugate of the local code frequency domain value, and then take the inverse Fourier transform, which is represented as:

$$[0125] \quad \begin{cases} R_1(n) = IFFT[FFT(s_{r1}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_q(n) = FFT[FFT(s_{rq}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_Q(n) = FFT[FFT(s_{rQ}(n)) * conj(FFT(c(n)))] \end{cases};$$

[0126] Among them, $FFT(\)$ represents fast Fourier transform of the received signals, $IFFT(\)$ represents the fast Fourier transform of the acquired signals, $conj(\)$ represents the complex conjugate of the acquired signal, and $c(n)$ represents the local C/A code;

[0127] The transfer function of the amplifier 4 provided with the sub-matched filter is:

$$H(z) = \sum_{i=0}^{N-1} h_i z^{-i} = \sum_{i=0}^{N-1} x_{N-1-i} Z^{-i} = X(z^{-1}) Z^{-(N-1)},$$

Ci is modulated by the hierarchical sequence u and v, u is the hierarchical Golay sequence $u=\{1, 1, 1, 1, 1, 1, -1, -1, 1, -1, 1, -1, 1, -1, -1, 1\}$, $v=\{1, 1, 1, -1, -1, 1, -1, -1, 1, 1, 1, -1, 1, -1, 1, 1\}$, $C_{16m+n}=unv_m$;

[0128] $H(z)=X(z)=C(z)=\sum_{i=0}^{L_u L_v-1} C_i z^{-i} = \sum_{i=0}^{L_u L_v-1} C_{16m+n} z^{-(16m+n)} = \sum_{i=1}^{L_u-1} u_n z^{-n} \sum_{i=1}^{L_v-1} v_m z^{-16m} = H(z_u)H(z_v)$, According to the hierarchical Golay sequence to improve the transfer function, there comes:

$$[0129] H(z_u)=[1+z^{-8}+z^{-1}(1-z^{-8})][1+z^{-4}+z^{-2}(1-z^{-4})];$$

$$[0130] H(z_v)=(1+z^{-1})[1-z^{-6}+z^{-8}+z^{-14}]+(1-z^{-1})[z^{-2}-z^{-4}+z^{-10}+z^{-12}];$$

[0131] An analyzer is electrically connected to the amplifier to compare the amplified signal with the corresponding computer data signal, and analyze the sound generated by the vibration frequency and the vibration signal of the current user at the current time;

[0132] The data compression method of the analyzer contains the following steps:

[0133] Step 1. When coding, first calculate the E1 value according to the formula

$$E_{1 \rightarrow 1} = E_{1n} + d_{n+1}, \text{ and then calculate the fitting residual according to } \Delta d'_n = \frac{4 \times d_n}{n} - \frac{6 \times E_{1n}}{n(n+1)}$$

and $DFR_n = \Delta d'_n - \lfloor \Delta d'_n \rfloor = \varepsilon$. When calculating these two, you need to make an overrun judgment on the result to determine whether E1 is over-limit, so as to avoid overflow caused by exceeding the upper limit of the sensor data bus. Determining whether the residual is over-limit is to achieve piecewise fitting;

[0134] Step 2. After all the fitting residuals of a piece of input data are calculated, there will be data packet generated in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$. Make entropy coding by the S-Huffman coding method, and send it out. For the decoding at the receiving end, a set of data for decoding will be received and the data packet shown in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$, will be restored. Calculate and restore all the original data according to the formula

$$d_{n-1} = d_n - \lfloor \Delta d'_n \rfloor - DFR_n = d_n - \left\lfloor \frac{4 \times d_n}{n} - \frac{6 \times E1_n}{n(n+1)} \right\rfloor - DFR_n$$

[0135] Furthermore, an amplifier 4 which includes:

[0136] A digitization module for digitizing the input signal;

[0137] The nonlinear compensation database module is connected with a digitization module, a output signal converter module, and a nonlinear compensation database module, and used to perform nonlinear digitization compensation processing on the input signal through the digitization module and the feedback signal through the nonlinear compensation database module;

[0138] The output signal converter module is connected with the nonlinear digital compensation module and the output circuit, and used to convert the input signal through the nonlinear digital compensation module and the signal fed back by the upper and lower electromagnetic coils into a signal suitable for system control;

[0139] The output circuit module connected with the output signal converter and the upper and lower electromagnetic coils to output the signals processed by the signal converter;

[0140] The position signal module is connected with the position sensor and the input interface module and used to process the position signal detected by the position sensor and transmit it to the input interface module;

[0141] The input interface is connected with the position signal module and the compensation parameter determination module and used to transmit the signal of the position signal module to the input interface of the compensation parameter determination module;

[0142] The compensation parameter determination module is connected with the input interface module, material parameter input module, and nonlinear compensation database module and used to comprehensively process the signal transmitted by the upper and lower coils and the position feedback signal transmitted by the input interface module according to the reference data provided by the material parameter input module;

[0143] The material parameter input module is connected with the compensation parameter determination module and used to provide and input the material parameter input module for the compensation parameter determination module;

[0144] Nonlinear compensation database module is connected with the nonlinear digitization compensation module and compensation parameter determination module and used to process the feedback signal transmitted by the compensation parameter determination module and input it into the nonlinear digital compensation module.

[0145] This invention provides a control method based on a network architecture detection control module built in the acquisition unit to perform control, which specifically includes:

[0146] There are decision indicators $p_1, p_2, \dots, p_m$ in the listening target and decision-making layer, and the network architecture layer under the listening target and decision-making layer has $C_1, C_2, \dots, C_N$ index sets, among which C_i contains the element $e_{i1}, e_{i2}, \dots, e_{in_i}, i = 1, 2, \dots, N$.

[0147] The control method is also characterized by taking the hearing target and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , that is, to construct a judgment matrix under the criterion p_s :

[0148]	e_{jk}	$e_{i1}, e_{i2}, \dots, e_{in_i}$	归一化特征向量	Normalized Feature Vector
	e_{i1}		$w_{i1}^{(jk)}$	
	e_{i2}		$w_{i2}^{(jk)}$	
	$\vdots$		$\vdots$	
	e_{in_i}		$w_{in_i}^{(jk)}$	

[0149] And get the weight vector by the characteristic root method $w_{i1}^{(jk)}, w_{i2}^{(jk)}, \dots, w_{in_i}^{(jk)}$.

[0150] The control method is also featured by

[0151] Repeating the above steps for $k=1, 2, \dots, n_i$ to obtain the matrix W_{ij} , which is shown below.

[0152]

$$W_{ij} = \begin{pmatrix} w_{i1}^{(j1)} & w_{i1}^{(j2)} & \dots & w_{i1}^{(jn_j)} \\ w_{i2}^{(j1)} & w_{i2}^{(j2)} & \dots & w_{i2}^{(jn_j)} \\ \vdots & \vdots & \vdots & \vdots \\ w_{in_i}^{(j1)} & w_{in_i}^{(j2)} & \dots & w_{in_i}^{(jn_j)} \end{pmatrix},$$

[0153] Of which, the column vector of W_{ij} is the ranking vector of the influence degree of the elements $e_{i1}, e_{i2}, \dots, e_{in_i}$ in C_i on the elements $e_{j1}, e_{j2}, \dots, e_{jn_j}$ in C_j . If the elements in C_j are not affected by the elements in C_i , $W_{ij}=0$.

[0154] The control method of the hearing detection control system as described in claim 3 is also characterized by repeatedly taking the listening target for $i=1, 2, \dots, N$; $j=1, 2, \dots, N$ and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , to obtain the super matrix W under the decision criterion p_s :

$$W = \begin{pmatrix} C_1 \begin{pmatrix} W_{11} & W_{12} & \cdots & W_{1N} \end{pmatrix} \\ C_2 \begin{pmatrix} W_{21} & W_{22} & \cdots & W_{2N} \end{pmatrix} \\ \vdots \\ C_N \begin{pmatrix} W_{N1} & W_{N2} & \cdots & W_{NN} \end{pmatrix} \end{pmatrix}$$

[0155]

[0156] The control method is also characterized by reflecting one-step dominance of the element i over the element j in the super matrix W . Calculating W^2 . The element W_{ij}^2 represents the two-step dominance of element i over element j . W^2 is still listed as a normalized matrix. Calculating W^3 , W^4 , ..., in the same manner. When W^∞ exists, the j th column of W^∞ is the limit relative weight vector of each element to j in the network architecture layer under the criterion p_s ,

$$W^\infty = \begin{pmatrix} W_{11}^\infty & W_{12}^\infty & \cdots & W_{1N}^\infty \\ W_{21}^\infty & W_{22}^\infty & \cdots & W_{2N}^\infty \\ \vdots & \vdots & \vdots & \vdots \\ W_{N1}^\infty & W_{N2}^\infty & \cdots & W_{NN}^\infty \end{pmatrix}$$

[0157]

[0158] Of which, the value of each row is the local weight vector of the corresponding element; when a row is all 0, the corresponding local weight is 1; arrange the local weights in the order of elements to obtain the local weight vector;

$$[0159] \quad Q = [q_{11}, \cdots, q_{1n_1}, q_{21}, \cdots, q_{2n_2}, \cdots, q_{N1}, \cdots, q_{Nn_N}]^T \circ$$

[0160] The control method of the hearing detection control system as described in claim 3 is characterized by the decision index which comprises the extremely small hearing value, the very small hearing value, the small hearing value, the relatively large hearing value, and the large hearing value.

[0161] For this invention, the sound signal is detected by the acquisition unit, and conducted through the penetration. The impurity sound is filtered out by the filter device. The normal sound signal is amplified by the amplifier, and analyzed and further screened and processed by the analyzer, and finally transmitted through the input converter and output converter to provide users with increased comfort.

[0162] The control method of the present invention can obtain more accurate hearing data.

[0163] The above are only preferred embodiments of the utility model patent and are not intended to limit this invention. Any modification, equivalent replacement, and improvement made within the spirit and principle of the invention shall fall under the protection scope of this invention.

Claims

1. A hearing detection control system, characterized by an acquisition unit for the detection of surrounding sound information;

The acquisition unit receives the returned audio signal by the microphone, and uses a band-pass filter to filter out unconcerned frequency bands. Use the window function to reduce the influence of spectrum leakage, take fast Fourier transform to obtain the frequency domain signal, and process it as decibel data and store it in the sampling queue Queue1;

Obtain the spectrum analysis window queue w1 from the sampling queue. The calculation formula is:

$$w_1(x) = Queue_1(x + [(f_0 - f_{shift}) * f_{min}]), x \in [0, 2 * [f_{shift} * f_{min}]] ;$$

Of which, Queue₁ is the sampling queue;

f₀ is the center frequency of the audio signal;

f_{shift} is the maximum value of frequency change;

f_{min} is the minimum resolution of the spectral signal;

Obtain the left mean value W_{left} and the right mean value W_{right} of the maximum frequency shift f_{shift} from the spectrum analysis window queue:

$$W_{left} = \frac{\sum_{i=0}^{[f_{shift} * f_{min}] - 2} w_1(i)}{[f_{shift} * f_{min}] - 2} ;$$

$$W_{right} = \frac{\sum_{i=[f_{shift} * f_{min}] + 2}^{2 * [f_{shift} * f_{min}]} w_1(i)}{[f_{shift} * f_{min}] - 2} ;$$

The cycle T_{simulate} of the simulation definition logic. If T_{simulate} is too long, the user will have an obvious sense of delay, and the human-computer interaction will be very poor. However, if T_{simulate} is too short, the sampling data cannot be updated in time, there will be a lot of wrong operations. According to the concept of definition, the inevitable error rate R_{error} is: due to the difference between the signal sampling frequency and the acquisition signal processing and analysis frequency, an inevitable error will occur.

The formula of T_{simulate} is:

$$T_{simulate} = \frac{T_0}{1 + R_{error}} ;$$

Of which, T_0 is the sampling period;

R_{error} is the inevitable error rate;

Define a threshold T_{Degree} that is affected by the energy E_{peak} of the center frequency f_0 and the weight α , calculate the change rate of the left mean value w_{left} and the right mean value w_{right} in a unit $T_{simulate}$ period, and compare it with the threshold T_{Degree} :

$$T_{Degree} = \frac{\alpha E_{peak}}{[f_{shift} * f_{min}]} \quad \alpha \in [0,2] ;$$

Of which, α is the weight;

E_{peak} is the energy of the center frequency f_0 ;

$$R_{left} = \frac{\Delta w_{left}}{T_0} ;$$

$$R_{right} = \frac{\Delta w_{right}}{T_0} ;$$

Comparison process:

$R_{left} > T_{Degree}$;

What happens is an action of moving away from the microphone, the message is recorded as L_{event} to the message queue $Queue_{event}$:

$R_{right} > T_{Degree}$;

What happens is an action of moving close to the microphone, the message is recorded as R_{event} to the message queue $Queue_{event}$:

$R_{left} > T_{Degree}$ and $R_{right} > T_{Degree}$;

What happens is a compound action (for example, a flip gesture, or back and forth movement of both hands), and the message is recorded as D_{event} to the message queue $Queue_{event}$;

Within the waiting time of T_{delay} , if a message is generated, the message is recorded to the message queue $Queue_{event}$, the timer is reset and wait for the T_{delay} time again; if no message is generated, a message processing analysis will be performed on the message queue $Queue_{event}$, and the queue will be cleared;

If there is no timer resetting within a certain T_{delay} time, there will be a message processing analysis on the message queue $\text{Queue}_{\text{event}}$ to obtain the message logic. Simulate the operation reflected by the message logic to achieve the control of the computer and complete the human-computer interaction;

The conductor is electrically connect to the acquisition unit to transmit sound information;

The filtering device is connect to the conductor to filter sound signals;

The filtering device performs down-conversion processing on the received intermediate frequency signal $r(n)$:

$$\begin{aligned} s_r(n) &= r(n) * e^{-j2\pi i * \Delta f} \\ &= \{h(l) * c(n) * e^{j*2\pi f_1} + P(n)\} * e^{-j2\pi i * \Delta f}; \end{aligned}$$

Among them, $s_r(n)$ is the baseband signal, $P(n)$ is the noise of the intermediate frequency signal $r(n)$ and the noise is complex Gaussian noise. The real and virtual parts obey the Gaussian distribution with mean value of 0 and the variance of δ^2 , f_1 is the signal frequency, Δf is the search step length, i is the number of searches. Divide the $s_r(n)$ signal into Q ms data represented as $s_{r1}(n), \dots, s_{rq}(n), \dots, s_{rQ}(n)$, and multiply each of them respectively with the complex conjugate of the local code frequency domain value, and then take the inverse Fourier transform, which is represented as:

$$\begin{cases} R_1(n) = IFFT[FFT(s_{r1}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_q(n) = FFT[FFT(s_{rq}(n)) * conj(FFT(c(n)))] \\ \dots \\ R_Q(n) = FFT[FFT(s_{rQ}(n)) * conj(FFT(c(n)))] \end{cases};$$

Among them, $FFT()$ represents fast Fourier transform of the received signals, $IFFT()$ represents the fast Fourier transform of the acquired signals, $conj()$ represents the complex conjugate of the acquired signal, and $c(n)$ represents the local C/A code;

An amplifier that is electrically connected to the filter device for amplifying the sound signal;

The transfer function of the amplifier provided with the sub-matched filter is:

$$H(z) = \sum_{i=0}^{N-1} h_i z^{-i} = \sum_{i=0}^{N-1} x_{N-1-i} Z^{-i} = X(z^{-1}) Z^{-(N-1)},$$

C_i is modulated by the hierarchical sequence u and v , u is the hierarchical Golay sequence $u=\{1, 1, 1, 1, 1, 1, -1, -1, 1, -1, 1, -1, 1, -1, -1, 1\}$, $v=\{1, 1, 1, -1, -1, 1, -1, -1, 1, 1, 1, -1, 1, -1, 1, 1\}$, $C_{16m+n}=u_nv_m$;

$$H(z)=X(z)=C(z)=\sum_{i=0}^{L_u L_v-1} C_i z^{-i}=\sum_{i=0}^{L_u L_v-1} C_{16m+n} z^{-(16m+n)}=\sum_{i=1}^{L_u-1} u_n z^{-n} \sum_{i=1}^{L_v-1} v_m z^{-16m}=H(z_u)H(z_v),$$

According to the hierarchical Golay sequence to improve the transfer function, there comes:

$$H(z_u)=[1+z^{-8}+z^{-1}(1-z^{-8})][1+z^{-4}+z^{-2}(1-z^{-4})];$$

$$H(z_v)=(1+z^{-1})[1-z^{-6}+z^{-8}+z^{-14}]+(1-z^{-1})[z^{-2}-z^{-4}+z^{-10}+z^{-12}];$$

An analyzer is electrically connected to the amplifier to compare the amplified signal with the corresponding computer data signal, and analyze the sound generated by the vibration frequency and the vibration signal of the current user at the current time;

The data compression method of the analyzer contains the following steps:

Step 1. When coding, first calculate the $E1$ value according to the formula

$$E1_{n+1} = E1_n + d_{n+1}, \text{ and then calculate the fitting residual according to } \Delta d'_n = \frac{4 \times d_n}{n} - \frac{6 \times E1_n}{n(n+1)}$$

and $DFR_n = \Delta d'_n - \lfloor \Delta d'_n \rfloor = \varepsilon$. When calculating these two, you need to make an overrun judgment on the result to determine whether $E1$ is over-limit, so as to avoid overflow caused by exceeding the upper limit of the sensor data bus. Determining whether the residual is over-limit is to achieve piecewise fitting;

Step 2. After all the fitting residuals of a piece of input data are calculated, there will be data packet generated in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$. Make entropy coding by the S-Huffman coding method, and send it out. For the decoding at the receiving end, a set of data for decoding will be received and the data packet shown in $\{d_n, E1_n, DFR_3, DFR_4, \dots DFR_n\}$, will be restored. Calculate and restore all the original data according to the formula;

$$d_{n-1} = d_n - \lfloor \Delta d'_n \rfloor - DFR_n = d_n - \left\lfloor \frac{4 \times d_n}{n} - \frac{6 \times E1_n}{n(n+1)} \right\rfloor - DFR_n$$

An input converter is electrically connected to the analyzer and used to convert an input signal into an electric input signal;

An output converter is electrically connected with the converter, and is used to convert the electric output signal into the audio output signal.

2. The hearing detection control system as described in claim 1 is characterized by an amplifier which includes a digitization module for digitizing the input signal;

The nonlinear compensation database module is connected with a digitization module, a output signal converter module, and a nonlinear compensation database module, and used to perform nonlinear digitization compensation processing on the input signal through the digitization module and the feedback signal through the nonlinear compensation database module;

The output signal converter module is connected with the nonlinear digital compensation module and the output circuit, and used to convert the input signal through the nonlinear digital compensation module and the signal fed back by the upper and lower electromagnetic coils into a signal suitable for system control;

The output circuit module connected with the output signal converter and the upper and lower electromagnetic coils to output the signals processed by the signal converter;

The position signal module is connected with the position sensor and the input interface module and used to process the position signal detected by the position sensor and transmit it to the input interface module;

The input interface is connected with the position signal module and the compensation parameter determination module and used to transmit the signal of the position signal module to the input interface of the compensation parameter determination module;

The compensation parameter determination module is connected with the input interface module, material parameter input module, and nonlinear compensation database module and used to comprehensively process the signal transmitted by the upper and lower coils and the

position feedback signal transmitted by the input interface module according to the reference data provided by the material parameter input module;

The material parameter input module is connected with the compensation parameter determination module and used to provide and input the material parameter input module for the compensation parameter determination module;

Nonlinear compensation database module is connected with the nonlinear digitization compensation module and compensation parameter determination module and used to process the feedback signal transmitted by the compensation parameter determination module and input it into the nonlinear digital compensation module.

3. A control method for a hearing detection control system as described in claim 1, is specially based on a network architecture detection control module built in the acquisition unit to perform control, which specifically includes:

There are decision indicators $p_1, p_2, \dots, p_m$ in the listening target and decision-making layer, and the network architecture layer under the listening target and decision-making layer has $C_1, C_2, \dots, C_N$ index sets, among which C_i contains the element $e_{i1}, e_{i2}, \dots, e_{in_i}, i = 1, 2, \dots, N$.

4. The control method of the hearing detection control system as described in claim 3 is also characterized by taking the hearing target and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , that is, to construct a judgment matrix under the criterion p_s :

e_{jk}	$e_{i1}, e_{i2}, \dots, e_{in_i}$	归一化特征向量	Normalized Feature Vector
e_{i1}		$w_{i1}^{(jk)}$	
e_{i2}		$w_{i2}^{(jk)}$	
$\vdots$		$\vdots$	
e_{in_i}		$w_{in_i}^{(jk)}$	

And get the weight vector by the characteristic root method $w_{i1}^{(jk)}, w_{i2}^{(jk)}, \dots, w_{in_i}^{(jk)}$ 。

5. The control method of the hearing detection control system as described in claim 3 is also featured by:

repeating the above steps for $k=1, 2, \dots, n_i$ to obtain the matrix W_{ij} , which is shown below.

$$W_{ij} = \begin{pmatrix} w_{i1}^{(j1)} & w_{i1}^{(j2)} & \dots & w_{i1}^{(jn_j)} \\ w_{i2}^{(j1)} & w_{i2}^{(j2)} & \dots & w_{i2}^{(jn_j)} \\ \vdots & \vdots & \ddots & \vdots \\ w_{in_i}^{(j1)} & w_{in_i}^{(j2)} & \dots & w_{in_i}^{(jn_j)} \end{pmatrix},$$

Of which, the column vector of W_{ij} is the ranking vector of the influence degree of the elements $e_{i1}, e_{i2}, \dots, e_{in_i}$ in C_i on the elements $e_{j1}, e_{j2}, \dots, e_{jn_j}$ in C_j . If the elements in C_j are not affected by the elements in C_i , $W_{ij}=0$.

6. The control method of the hearing detection control system as described in claim 3 is also characterized by repeatedly taking the listening target for $i=1, 2, \dots, N$; $j=1, 2, \dots, N$ and the decision-making index p_s ($s=1, 2, \dots, m$) as the criterion and the element e_{jk} ($k=1, 2, \dots, n_j$) in C_j as the sub-criterion, and comparing the influence of the index set C_i on e_{jk} , to obtain the super matrix W under the decision criterion p_s :

$$W = \begin{pmatrix} C_1 & \begin{pmatrix} W_{11} & W_{12} & \dots & W_{1N} \end{pmatrix} \\ C_2 & \begin{pmatrix} W_{21} & W_{22} & \dots & W_{2N} \end{pmatrix} \\ \vdots & \vdots \\ C_N & \begin{pmatrix} W_{N1} & W_{N2} & \dots & W_{NN} \end{pmatrix} \end{pmatrix}$$

7. The control method of the hearing detection control system as described in claim 3 is also characterized by reflecting one-step dominance of the element i over the element j in the super matrix W . Calculating W^2 . The element W_{ij}^2 represents the two-step dominance of element i over element j . W^2 is still listed as a normalized matrix. Calculating $W^3, W^4, \dots$, in the same

manner. When W^∞ exists, the j th column of W^∞ is the limit relative weight vector of each element to j in the network architecture layer under the criterion p_s ,

$$W^\infty = \begin{pmatrix} W_{11}^\infty & W_{12}^\infty & \cdots & W_{1N}^\infty \\ W_{21}^\infty & W_{22}^\infty & \cdots & W_{2N}^\infty \\ \vdots & \vdots & \vdots & \vdots \\ W_{N1}^\infty & W_{N2}^\infty & \cdots & W_{NN}^\infty \end{pmatrix}$$

Of which, the value of each row is the local weight vector of the corresponding element; when a row is all 0, the corresponding local weight is 1;

Arrange the local weights in the order of elements to obtain the local weight vector;

$$Q = [q_{11}, \cdots, q_{1n_1}, q_{21}, \cdots, q_{2n_2}, \cdots, q_{N1}, \cdots, q_{Nn_N}]^T \circ$$

8. The control method of the hearing detection control system as described in claim 3 is characterized by the decision index which comprises the extremely small hearing value, the very small hearing value, the small hearing value, the relatively large hearing value, and the large hearing value.