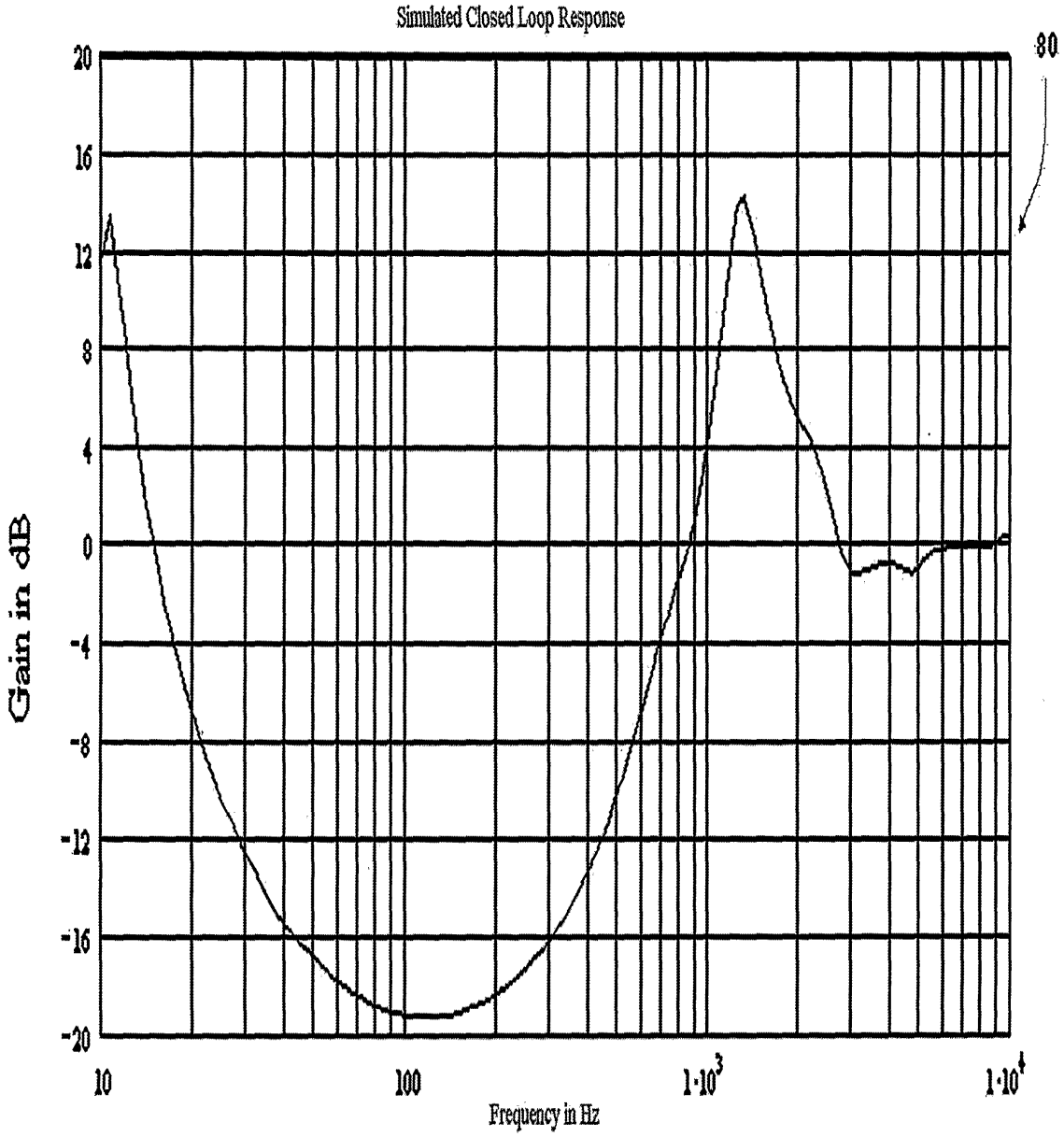

Simulated Closed Loop Response
80
20
16
12
8
4
0
-4
-8
-12
-16
-20
Gain in dB
10
100
1·10^3
1·10^4
Frequency in Hz

Fig. 8

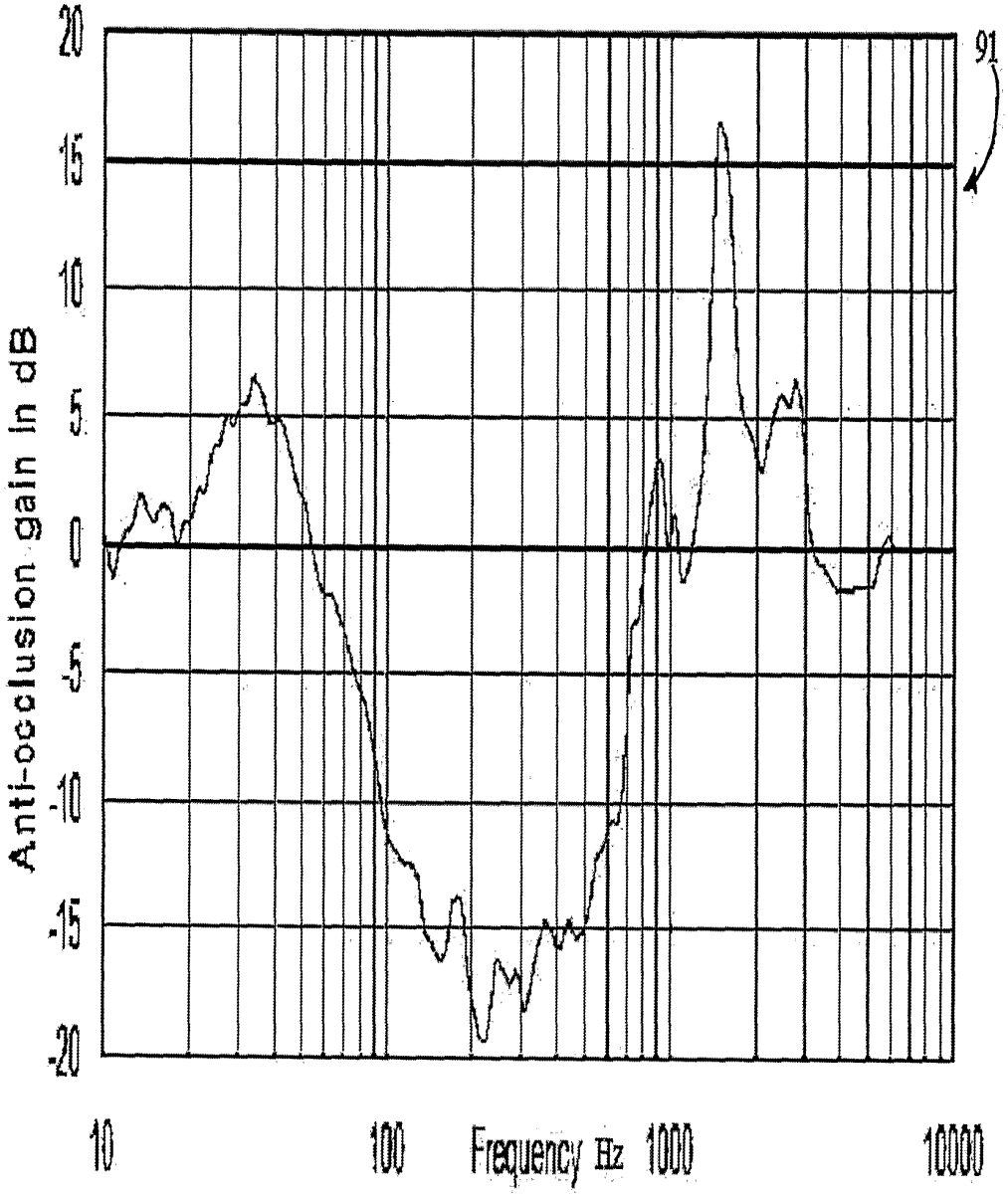

20
15
10
5
0
-5
-10
-15
-20
Anti-occlusion gain in dB
91
10
100
Frequency Hz
1000
10000

Fig. 9

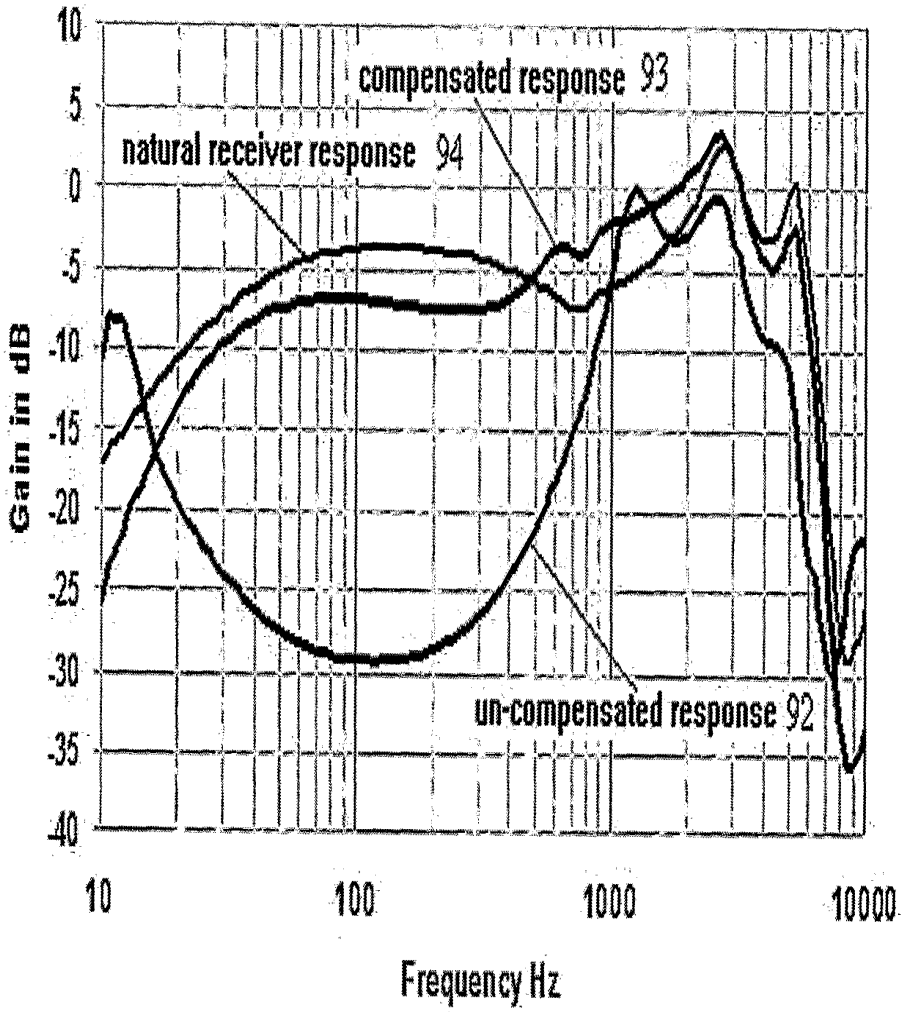

compensated response 93
natural receiver response 94
un-compensated response 92
Gain in dB
10
5
0
-5
-10
-15
-20
-25
-30
-35
-40
10
100
1000
10000
Frequency Hz

Fig. 10

External
Microphone
Main Hearing Aid Processor
104
Receiver
5
103
Tube
Wax protection
A
D
C
B
110
108
fadepate
tube
109
111
7
Battery
105
Optional
small vent
Wax protection
Earmold
Internal
Microphone
100
102
101
100

Fig. 11

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

A. CLASSIFICATION OF SUBJECT MATTER

Int. Cl. [7]: H04R 29/00; A61B 5/00

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)

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

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
WPAT: hearing aid, occlusion, gain, frequency range, sound level, signal strength and similar terms.
USPTO: hearing aid, occlusion, reduce

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 6687377 B2 (VOIX et al.) 3 February 2004 Col. 3 line 33-Col. 6 line 22	1-24
A	WO 2002/085063 A2 (WIDEX A/S et al.) 24 October 2002 Page 4 line 14-Page 10 line 32	1-24
A	US 6766031 B1 (WESTERMANN) 20 July 2004 Col. 2 line 18-Col. 5 line 28	1-24
A	CA 2381516 A1 (GENNUM CORPORATION, CA) 12 October 2002 Page 1 line 20-Page 18 line 2	1-24

[X] Further documents are listed in the continuation of Box C [X] See patent family annex

* Special categories of cited documents:

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

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

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

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

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 November 2005	9 NOV 2005
Name and mailing address of the ISA/AU AUSTRALIAN PATENT OFFICE PO BOX 200, WODEN ACT 2606, AUSTRALIA E-mail address: pct@ipaustralia.gov.au Facsimile No. (02) 6285 3929	Authorized officer **JUZER KHANBHAI** Telephone No : (02) 6283 2176

Form PCT/ISA/210 (second sheet) (January 2004)

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5195139 A (GAUTHIER) 16 March 1993 Col. 1 line 39-Col. 3 line 22	1-24

INTERNATIONAL SEARCH REPORT

International application No.
PCT/AU2005/001506

This Annex lists the known "A" publication level patent family members relating to the patent documents cited in the above-mentioned international search report. The Australian Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

Patent Document Cited in Search Report		Patent Family Member					
US	6687377	AU	20394/02	CA	2477010	CN	1483298
		EP	1356255	US	2002076057	WO	0250499
		ZA	200305523				
WO	2002/085063	CA	2442538	EP	1250027	US	2002150269
US	6766031	AU	69189/98	CA	2286037	DK	42297
		EP	0980641	WO	9847318		
CA	2381516	N/A					
US	5195139	AU	20280/92	EP	0584259	WO	9221218

Due to data integration issues this family listing may not include 10 digit Australian applications filed since May 2001.

END OF ANNEX

(19)日本国特許庁(JP)　(12)公開特許公報(A)　(11)特許出願公開番号
特開2005-168888
(P2005-168888A)
(43)公開日　平成17年6月30日(2005.6.30)

(51)Int.Cl.[7]	F I			テーマコード（参考）
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G10L 15/00	A63F	5/04	512C	
G10L 15/28	G10L	3/00	551H	
	G10L	3/00	511	

審査請求 未請求　請求項の数 8　OL　（全 17 頁）

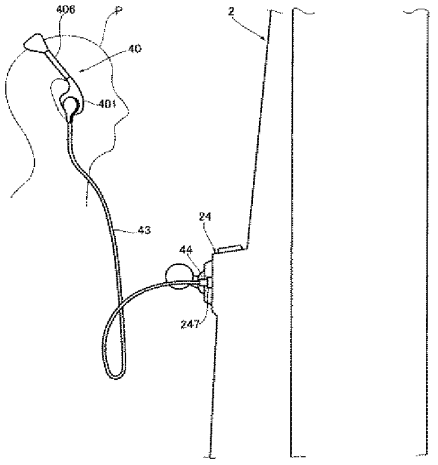
(21)出願番号　特願2003-414772 (P2003-414772)
(22)出願日　平成15年12月12日 (2003.12.12)

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Ｆターム(参考)　5D015 BB01　KK01

(54)【発明の名称】回胴式遊技機及びその操作制御方法

(57)【要約】
【課題】　遊技操作における手の疲労を抑制する等の回胴式遊技機及びその操作制御方法を提供する。
【解決手段】　リールを複数有する回胴式遊技機であって、骨伝導マイクを有するヘッドセットと、ヘッドセットを装着する遊技者が発声し骨伝導マイクを介して得た信号から単語情報を音声認識する手段と、この単語情報が管理テーブルに予め登録されている操作命令語と一致した場合に、その操作命令語に対応づけられた遊技動作を行う出力手段を備える。本発明によれば、音声入力による遊技操作を可能にし、遊技操作において遊技者の手への負担を軽減する。
【選択図】　図３



【特許請求の範囲】
【請求項1】
遊技者による操作に応じて周面に描かれた図柄を変動表示するリールを複数有する回胴式遊技機であって、
骨伝導マイクを有し、遊技者に装着されるヘッドセットと、
前記骨伝導マイクにより検出された信号に含まれている単語情報を音声認識する音声認識処理手段と、
前記リールの回転を開始させる又は回転を停止させる遊技動作に対し操作命令語が対応づけられて登録されている管理テーブルを検索し、音声認識された当該単語情報が前記管理テーブルに登録されている前記操作命令語のひとつと一致する場合に、当該一致する操作命令語に対応づけられている前記遊技動作を行うための電気的なスイッチ信号を出力する出力手段と、を備えることを特徴とする回胴式遊技機。
【請求項2】
前記ヘッドセットは、スピーカを更に備えることを特徴とする請求項１に記載の回胴式遊技機。
【請求項3】
前記ヘッドセットは、フロントパネルに設けられたコネクタを介して当該回胴式遊技機の本体に接続可能であることを特徴とする請求項１又は２に記載の回胴式遊技機。
【請求項4】
前記ヘッドセットが前記フロントパネルに接続されているとき、これを検知することで、遊技者の音声に応じて当該回胴式遊技機を操作する操作モードへ自動的に切り換える操作モード切換手段を更に備えることを特徴とする請求項３に記載の回胴式遊技機。
【請求項5】
前記操作命令語は、一の前記遊技動作に対して複数、前記管理テーブルに登録可能であることを特徴とする請求項１～４の何れか１項に記載の回胴式遊技機。
【請求項6】
音声認識された前記単語情報が前記操作命令語のひとつと一致したとき、遊技者に報知する報知手段を更に備えることを特徴とする請求項１～５の何れか１項に記載の回胴式遊技機。
【請求項7】
前記報知手段は、ストップボタンに内蔵されるランプであって、該ランプを点灯する又は点灯色を変化させることで前記報知を行うことを特徴とする請求項６に記載の回胴式遊技機。
【請求項8】
遊技者による操作に応じて周面に描かれた図柄を変動表示するリールを複数有する回胴式遊技機における操作制御方法であって、
骨伝導マイクを備えるヘッドセットを装着した遊技者により発声された音声を前記骨伝導マイクが検出し、その音声に含まれている単語情報を音声認識するステップと、
前記リールの回転を開始させる又は回転を停止させる遊技動作に対し操作命令語が対応づけられて登録されている管理テーブルを検索することで、音声認識された当該単語情報と前記管理テーブルに予め登録されている前記操作命令語とを比較照合するステップと、
当該単語情報が前記操作命令語のひとつと一致した場合に、当該一致した操作命令語に対応づけられている前記遊技動作を行うステップと、を備えることを特徴とする操作制御方法。

【発明の詳細な説明】
【技術分野】
【0001】
本発明は、回胴装置を備えるスロットマシン等の回胴式遊技機に関し、特に遊技操作における手の疲労を抑制する等の回胴式遊技機及びその操作制御方法に関するものである。
【背景技術】

【0002】
従来、スロットマシン等の回胴式遊技機においては、遊技者に面するフロントパネルに遊技操作をするためのスイッチまたはレバー等の操作入力手段が設けられ、これら操作入力手段を遊技者が手で直接的に操作することでゲームが進行するように構成されている。
【0003】
一般的なスロットマシンは、例えば図１に示されるように、３個のリール３１１ａ，３１１ｂ，３１１ｃから形成された回胴装置と、メダル投入部２４１と、スタートレバー２４５と、ストップボタン２４６ａ，２４６ｂ，２４６ｃ等を備え、これらがフロントパネル２の適宜の位置に配置されている。
【0004】
円筒状の各リール３１１ａ，３１１ｂ，３１１ｃの周面には図示しない複数種類の図柄が描かれており、これらの図柄がパネル面２２１に形成されている透明な表示窓２２２を通して目視されるようになっている。スタートレバー２４５は、傾倒操作されることでオン動作するレバースイッチであり、ストップボタン２４６ａ，２４６ｂ，２４６ｃは、押圧操作されることでオン操作する押しボタン式のスイッチである。
【0005】
従来のスロットマシンは、遊技者によってメダル投入部２４１に遊技メダルが投入されるとゲームの開始が有効となり、遊技者側に若干突き出た操作卓２４に配置されているスタートレバー２４５が傾倒操作されると、全てのリール３１１ａ，３１１ｂ，３１１ｃが一斉に回転する。そして、遊技者が何れか任意のストップボタン２４６ａ，２４６ｂ，２４６ｃを押圧操作すると、そのストップボタンに対応するリールが制動されて停止する。そして、全てのリール３１１ａ，３１１ｂ，３１１ｃが停止操作され、それらが表示する図柄が所定の組合せで揃うと入賞となり、入賞の種類に応じた数の遊技メダルが排出口２３２から払い出されるようになっている。
【発明の開示】
【発明が解決しようとする課題】
【0006】
しかし、上述した従来のスロットマシンにおいては、遊技者は１回のゲームにつき少なくとも３度ストップボタン３１１ａ，３１１ｂ，３１１ｃを押圧操作しなければならず、長時間にわたってゲームを続けると遊技者の手に疲労がたまるという問題があった。
【0007】
また、手が不自由な障害者、または左利きの遊技者であって、上述のスタートレバー２４５と同じ側の手でストップボタン３１１ａ，３１１ｂ，３１１ｃも操作する等の遊技者にとっては、ゲーム操作における手への負担が大きくなり、数回のゲームであっても著しく手が疲労してしまうという問題もあった。
【0008】
本発明はこうした従来の問題に鑑みてなされたものであり、遊技操作における手の疲労を抑制し、遊技の操作性又は利便性等を高める等の回胴式遊技機及びその操作制御方法を提供することを目的としている。
【課題を解決するための手段】
【0009】
請求項１に記載の発明は、遊技者による操作に応じて周面に描かれた図柄を変動表示するリールを複数有する回胴式遊技機であって、骨伝導マイクを有し、遊技者に装着されるヘッドセットと、前記骨伝導マイクにより検出された信号に含まれている単語情報を音声認識する音声認識処理手段と、前記リールの回転を開始させる又は回転を停止させる遊技動作に対し操作命令語が対応づけられて登録されている管理テーブルを検索し、音声認識された当該単語情報が前記管理テーブルに登録されている前記操作命令語のひとつと一致する場合に、当該一致する操作命令語に対応づけられている前記遊技動作を行うための電気的なスイッチ信号を出力する出力手段と、を備えることを特徴とする。
【0010】

請求項１に記載の回胴式遊技機によれば、ヘッドセットを装着した遊技者がリールの回転または停止等の操作を行う際、適宜の単語を発声すると、音声認識処理手段は、骨伝導マイクを介して検出した信号からこれに含まれている単語情報を音声認識する。出力手段は、リールの回転又は停止等の各遊技動作に対しそれぞれ操作命令語が対応づけられて登録されている管理テーブルを検索し、音声認識された当該単語情報が、登録されている操作命令語のひとつと一致した場合、一致した当該操作命令語に対応づけられている遊技動作を行うための電気的なスイッチ信号を出力する。

【0011】

すなわち、ヘッドセットを装着した遊技者がリールを回転または停止させるための予め登録されている操作命令語を発声することで、当該操作命令語に対応する遊技動作が行われる。これにより、音声による操作を可能にする。

【0012】

また、ヘッドセットに備えられた骨伝導マイクは、当該ヘッドセットを装着している遊技者の音声のみを検出する。つまり、他の遊技者の音声や周囲の環境音等の雑音を検出することなく、これら雑音による誤作動を防止する。

【0013】

請求項２に記載の発明は、請求項１に記載の回胴式遊技機であって、前記ヘッドセットは、スピーカを更に備えることを特徴とする。

【0014】

請求項２に記載の回胴式遊技機によれば、スピーカを介して当該回胴遊技機の遊技を演出する効果音等を、ヘッドセットを装着している遊技者に供給する。

【0015】

請求項３に記載の発明は、請求項１又は２に記載の回胴式遊技機であって、前記ヘッドセットは、フロントパネルに設けられたコネクタを介して当該回胴式遊技機の本体に接続可能であることを特徴とする。

【0016】

請求項３に記載の回胴式遊技機によれば、ヘッドセットは、フロントパネルに設けられたコネクタを介して当該回胴式遊技機の本体に接続される。音声操作する場合にはヘッドセットはフロントパネルに接続され、通常の手操作をする場合には、フロントパネルから外される。これにより、本回胴式遊技機は、これら２つの操作モードを使い分けることができる。

【0017】

請求項４に記載の発明は、請求項３に記載の回胴式遊技機であって、前記ヘッドセットが前記フロントパネルに接続されているとき、これを検知することで、遊技者の音声に応じて当該回胴式遊技機を操作する操作モードへ自動的に切り換える操作モード切換手段を更に備えることを特徴とする。

【0018】

請求項４に記載の回胴式遊技機によれば、ヘッドセットがフロントパネルに接続されているとき、操作モード切換手段はこれを検知し、遊技者の音声によって操作を行う操作モードへ自動的に切り換える。これにより、本回胴式遊技機は、これら２つの操作モードを使い分けることができ、遊技者の利便性も向上する。

【0019】

請求項５に記載の発明は、請求項１～４の何れか１項に記載の回胴式遊技機であって、前記操作命令語は、一の前記遊技動作に対して複数、前記管理テーブルに登録可能であることを特徴とする。

【0020】

請求項５に記載の回胴式遊技機によれば、一の遊技動作に対して複数の操作命令語を管理テーブルに登録することが可能である。つまり、遊技者が好む又は覚えやすい等の任意の操作命令語を管理テーブルに登録することで、音声操作における操作性又は遊技者への利便性を向上させる。

【0021】
請求項６に記載の発明は、請求項１～５の何れか１項に記載の回胴式遊技機であって、音声認識された前記単語情報が前記操作命令語のひとつと一致したとき、遊技者に報知する報知手段を更に備えることを特徴とする。
【0022】
請求項６に記載の回胴式遊技機によれば、報知手段は、遊技者が発声し音声認識された単語が所定の操作命令語であると判定されたとき、遊技者に報知する。これにより、遊技者は音声入力した当該操作が有効なものとして受付けられたことを確認することができ、音声操作における利便性が向上する。
【0023】
請求項７に記載の発明は、請求項６に記載の回胴式遊技機であって、前記報知手段は、ストップボタンに内蔵されるランプであって、該ランプを点灯する又は点灯色を変化させることで前記報知を行うことを特徴とする。
【0024】
請求項７に記載の回胴式遊技機によれば、遊技者が発声し音声認識された単語が所定の操作命令語であると判定されたとき、ストップボタンに内蔵されたランプを点灯又はその点灯色を変化させることで遊技者に報知する。これにより、遊技者は音声入力した当該操作が有効なものとして受付けられたことを確実に知ることができ、音声操作における利便性が向上する。
【0025】
請求項８に記載の発明は、遊技者による操作に応じて周面に描かれた図柄を変動表示するリールを複数有する回胴式遊技機における操作制御方法であって、骨伝導マイクを備えるヘッドセットを装着した遊技者により発声された音声を前記骨伝導マイクが検出し、その音声に含まれている単語情報を音声認識するステップと、前記リールの回転を開始させる又は回転を停止させる遊技動作に対し操作命令語が対応づけられて登録されている管理テーブルを検索することで、音声認識された当該単語情報と前記管理テーブルに予め登録されている前記操作命令語とを比較照合するステップと、当該単語情報が前記操作命令語のひとつと一致した場合に、当該一致した操作命令語に対応づけられている前記遊技動作を行うステップと、を備えることを特徴とする。
【0026】
請求項８に記載の操作制御方法によれば、ヘッドセットを装着した遊技者がリールの回転または停止等の操作を行う際、適宜の単語を発声すると、骨伝導マイクを介して検出した信号からこれに含まれている単語情報を音声認識する。そして、リールの回転又は停止等の各遊技動作に対しそれぞれ操作命令語が対応づけられて登録されている管理テーブルを検索することで、音声認識された当該単語情報と登録されている操作命令語とを比較照合する。音声認識された単語情報が登録されている操作命令語のひとつと一致した場合に、当該一致した操作命令語に対応づけられている遊技動作を実行する。
【0027】
すなわち、ヘッドセットを装着した遊技者がリールの回転を開始または停止させるための予め登録されている操作命令語を発声することで、当該操作命令語に対応する遊技動作が行われる。これにより、音声による操作を可能にする。
【発明の効果】
【0028】
請求項１に記載の発明によれば、ヘッドセットを装着した遊技者がリールを回転または停止させるための予め登録されている操作命令語を発声することで、当該操作命令語に対応する遊技動作が行われる。これにより、遊技操作における手の負担を軽減し、手の疲労を抑制することができる。
【0029】
また、ヘッドセットに備えられた骨伝導マイクは、当該ヘッドセットを装着している遊技者の音声のみを検出する。つまり、他の遊技者の音声や周囲の環境音等の雑音は検出さ

れないので、これら雑音による誤作動を防止する。

【0030】

請求項２に記載の発明によれば、スピーカを介して当該回胴遊技機の遊技を演出する効果音等を、ヘッドセットを装着している遊技者に供給する。これにより、遊技者は、周囲の環境音に干渉されず遊技の演出効果音等を楽しむことができ、遊技の興趣が高まる。

【0031】

請求項３に記載の発明によれば、ヘッドセットをフロントパネルから着脱させることで、音声操作する場合と通常の手操作をする場合の２つの操作モードを使い分けることができる。

【0032】

請求項４に記載の発明によれば、ヘッドセットをフロントパネルから着脱させることで、音声操作する操作モードと通常の手操作の操作モードとが自動的に切り換わり、遊技者の利便性が向上する。

【0033】

請求項５に記載の発明によれば、遊技者が好む又は覚えやすい等の任意の操作命令語を登録することが可能なので、音声操作における操作性又は遊技者への利便性を向上させることができる。

【0034】

請求項６及び７に記載の発明によれば、遊技者は、音声入力した操作が有効なものとして受付けられたことを確認することができる。これにより、音声操作における操作性又は利便性を向上させることができる。

【0035】

請求項８に記載の発明によれば、ヘッドセットを装着した遊技者がリールを回転または停止させるための予め登録されている操作命令語を発声することで、当該操作命令語に対応する遊技動作が行われる。これにより、遊技操作における手の負担を軽減し、手の疲労を抑制することができる。

【発明を実施するための最良の形態】

【0036】

以下、本発明の回胴式遊技機における好適な一実施形態としてスロットマシン１を説明する。ここで、図２は、スロットマシン１のフロントドア２の構成を表す正面図である。

【0037】

フロントドア２は、上部パネル部２１と、中部パネル部２２と、下部パネル部２３とを備え、これらは遊技者側に面する化粧板としてデザインされ硬質プラスチックにより一体的に形成されている。また、中部パネル部２２と下部パネル部２３との間には、遊技者側に若干突き出して操作卓２４が一体的に形成されており、スロットマシン１のゲームの操作を行うための後述するスイッチが複数配置されている。

【0038】

上部パネル部２１の中央には、ゲームを演出する映像を表示するための液晶表示装置２１１が配置され、透明な硬質プラスチック板で形成された表示窓２１２を通して液晶表示装置２１１の映像が目視されるようになっている。表示窓２１２の両側には、複数のスリット孔からなる放音部２１４ａ，２１４ｂが形成され、ゲームを演出する効果音を発するための図示しないスピーカ（２５１ａ，２５１ｂ）が放音部２１４ａ，２１４ｂの裏面に取り付けられている。また、上部パネル部２１の上端部分には、高輝度の発光ダイオードを内蔵しこれらを点灯または点滅させることで光による演出を行うための演出用ランプ２１３が配設されている。

【0039】

中部パネル部２２には、硬質プラスチック板等で形成されたパネル面２２１が設けられている。パネル面２２１のほぼ中央には略長方形の透明な表示窓２２２が形成され、表示窓２２２を通して筐体３内に設けられている回胴装置３１の３個のリール３１１ａ，３１１ｂ，３１１ｃを目視できるようになっている。

【0040】
ここで、各リール３１１ａ，３１１ｂ，３１１ｃは、それぞれ直径が同一の円筒形状を有しており、回転軸に沿って３個並べて配列されている。そして、各リール３１１ａ，３１１ｂ，３１１ｃの外周面にはその周方向に沿って複数種類の図柄（図示略）が描かれており、遊技者はパネル面２２１に形成されている表示窓２２２を通して一のリールに描かれた上下方向３個の図柄と、横方向に配列されたリール３列の図柄を目視することができるようになっている。また、遊技者の操作に応じて、各リール３１１ａ，３１１ｂ，３１１ｃは回転及び停止の動作が行われ、これにより、表示されるリールの図柄の種類が変更されるようになっている。
【0041】
また、中部パネル部２２のパネル面２２１には、例えば入賞役への内部当選など遊技状態に関する情報をランプ等で点灯表示する遊技情報表示装置２２３と、複数の発光ダイオードがマトリクス状に配列され各発光ダイオードを選択的に発光させることでドット画像を演出表示するドット表示装置２２４と、スロットマシン１にクレジット（貯留）されている遊技メダル（以下、単に「メダル」）数や、入賞によって獲得したメダル数、または入賞役への当選回数等の数値情報を表示する７セグメントＬＥＤからなる数値情報表示装置２２５が設けられている。
【0042】
更に中部パネル部２２の両側部には、高輝度発光ダイオードを内蔵する演出用ランプ２２６ａ，２２６ｂが配設されている。
【0043】
中部パネル部２２の下側には、遊技者側（前方）に向かって若干下がりながら傾斜した平面を有する上述の操作卓２４が形成されている。操作卓２４の当該平面の右側にはメダルを投入するためのメダル投入部２４１が設けられ、当該平面の左側には押しボタンスイッチである３個のベットボタン２４２，２４３，２４４が設けられている。
【0044】
なお、図示はしていないが、メダル投入部２４１には投入されたメダルが正規のものかどうか判別するセレクト機構が筐体３内に連設されており、正規のメダルが投入された場合にこれを検出し検出信号を後述の主制御基板（３００）へ送出するメダル投入センサ（２４８）がセレクト機構に組み込まれている。
【0045】
ベットボタン２４２，２４３，２４４は、スロットマシンの１ゲームに賭けるメダルの枚数を提示するためのボタンスイッチである。ゲームを開始する際に、ベットボタン２４２が押圧操作されることで、貯留されているメダルから１枚のメダルが当該ゲームに対して賭けられる。同様に、ベットボタン２４３または２４４が押圧操作されることで、２枚または３枚のメダルが賭けられる。なお、ベットボタン２４４は、１回のゲームに賭けることができる最多枚数のメダルを提示するために設けられていることから、特に「マックスベットボタン」と呼ばれている。
【0046】
操作卓２４の前面左側には、リール３１１ａ，３１１ｂ，３１１ｃの回転開始を指示するためのスタートレバー２４５が設けられている。スタートレバー２４５は、先端に球形の操作ノブを有する揺動可能な操作旱を備え、操作旱が傾倒操作されるとオン、操作旱から手が離されるとスプリングの付勢力によって自動的に元の位置に戻ってオフ状態となるスイッチユニットで形成されている。
【0047】
操作卓２４の前面ほぼ中央には、リール３１１ａ，３１１ｂ，３１１ｃの回転停止を指示するための押しボタンスイッチである３個のストップボタン２４６ａ，２４６ｂ，２４６ｃが各リール３１１ａ，３１１ｂ，３１１ｃに対応して設けられている。ストップボタン２４６ａ，２４６ｂ，２４６ｃは、押圧操作することで回転中のリール３１１ａ，３１１ｂ，３１１ｃをそれぞれ独立して停止させるためのスイッチである。また、各ストップ

ボタン２４６ａ，２４６ｂ，２４６ｃは、青色及び赤色を発光表示する２個の発光ダイオードをそれぞれ内蔵している。

【0048】

操作卓２４の前面右側には、遊技者が音声を入力することでリール３１１ａ，３１１ｂ，３１１ｃの回転を開始または停止させる遊技操作を行う後述の骨伝導マイクを有するヘッドセット４０を接続するためのコネクタソケット２４７が配設されている。

【0049】

下部パネル部２３には、スロットマシン１のゲーム内容または登場キャラクタなどを紹介するためのパネル２３１が設けられている。パネル２３１の下側には、メダルを排出するための排出口２３２と、排出されたメダルを受ける受皿２３３とが設けられている。また排出口２３２に隣接して、多数の孔から形成された放音部２３４ａ，２３４ｂが設けられ、演出効果音を発する図示しないスピーカ（２５２ａ，２５２ｂ）が放音部２３４ａ，２３４ｂの裏面に配置されている。

【0050】

本スロットマシン１は、遊技者がメダル投入部２４１よりメダルを投下し、何れかのベットボタン２４２，２４３，２４４を押圧操作するとゲームの開始が有効となる。通常の手操作においては、遊技者によりスタートレバー２４５が傾倒操作されると、各リール３１１ａ，３１１ｂ，３１１ｃが一斉に回転する。各リール３１１ａ，３１１ｂ，３１１ｃが所定の回転数に達し、遊技者が任意の順序でストップボタン２４６ａ，２４６ｂ，２４６ｃを押圧操作することにより対応するリールを停止させる。そして、全て停止した３個のリール３１１ａ，３１１ｂ，３１１ｃにより表示される図柄が所定の組合せで揃うと入賞となり、その入賞の種類に応じた枚数のメダルが遊技者に配当されるようになっている。

【0051】

本スロットマシン１の操作においては、上述のストップボタン３１１ａ，３１１ｂ，３１１ｃ等を使った操作に加え、遊技者が発声する所定の単語（ここでは、「操作命令語」と呼んでいる）を音声入力して各リールに対して回転及び停止の動作を行う操作機能を有している。次に、この機能を実現するための一の構成要素である、骨伝導マイク（４１）を備えるヘッドセット４０を説明する。

【0052】

図３は、スロットマシン１側のコネクタソケット２４７に接続されたヘッドセット４０の構成を表した側面図である。

【0053】

図３において、ヘッドセット４０は、略Ｕ字状に形成され内側に向けて弾性付勢されたヘッドバンド４０６と、ヘッドバンド４０６の両端部分に設けられた耳あて部４０１とから主に構成されている。各耳あて部４０１には後述の小型のスピーカ（４２）が設けられ、また少なくとも一方の耳あて部４０１に後述の骨伝導マイク（４１）が設けられている。弾性付勢されたヘッドバンド４０６で遊技者Ｐの頭部を挟みながらヘッドセット４０の耳あて部４０１を遊技者Ｐの耳に当接させることで、ヘッドセット４０は遊技者Ｐに装着される。

【0054】

ヘッドセット４０は、内蔵する上述のスピーカ（４２）及び骨伝導マイク（４１）の信号線等を被覆するコード４３の先端部分に設けられたコネクタプラグ４４を、操作卓２４に配設されているコネクタソケット２４７に差し込むことで、スロットマシン１本体に接続され、また取り外すことも可能となっている。

【0055】

なお、ヘッドセット４０のコネクタプラグ４４がスロットマシン１のコネクタソケット２４７に接続されると、スロットマシン１本体側に設けられている操作モード切換回路（４４３）がこれを検出し、スロットマシン１の全体の動作を制御する後述の主制御基板（３００）は、遊技者による各リール３１１ａ，３１１ｂ，３１１ｃに対する操作をストッ

プボタン２４６ａ，２４６ｂ，２４６ｃ等の手操作によるものからヘッドセット４０に設けられている骨伝導マイク４１を使った音声操作によるものに自動的に切り換えるように構成されている。

【0056】

また、操作モード切換回路（４４３）がヘッドセット４０の接続状態を検出する方法としては、コネクタソケット２４７内に設けられ、コネクタプラグ４４が差し込まれたときに機械的に動作するスイッチを利用する方法、またはコネクタソケット２４７の端子電極間の抵抗（インピーダンス）を監視し、その違いから接続状態を検出する方法等、適宜の方法を採用することができる。

【0057】

図４は、骨伝導マイク４１が備えられた耳あて部４０１の構造を表す断面図である。同図において、耳あて部４０１の本体は、樹脂等により一体的に形成され、ユニット化された小型のスピーカ４２が耳あて部４０１の本体の内室に設置されている。スピーカ４２の前面側に前記内室を塞ぐ保護板４０４が配置されている。保護板４０４には、貫通する複数の孔４０５が形成され、これらの孔４０５を通してスピーカ４２からの鳴動音が放出される。保護板４０４は、耳あて部４０１本体の内側に形成されている環状縁部４０２に固着する保護メッシュ４０３で被覆されている。

【0058】

また、保護板４０４の略中央にユニット化された骨伝導マイク４１が取り付けられている。

【0059】

骨伝導マイクは、人が発声することにより生じる微弱振動（骨伝導音）を人体の一部を介して検出し電気的な音信号に変換するものである。本実施形態による骨伝導マイク４１は、遊技者の耳の外耳道に接触しその遊技者の発声に起因する微弱振動を検出する接触子４１２と、接触子４１２内に設けられ接触子の振動を電気信号として出力する圧電素子４１１と、圧電素子４１１の出力信号を所定レベルの電圧又は電流信号に増幅等する信号回路基板４１３とを備えている。

【0060】

かかる構成のヘッドセット４０によれば、遊技者Ｐに装着されスロットマシン１の本体に接続されると、遊技者Ｐが発声することで生じる骨伝導音を遊技者Ｐの外耳道に接触する骨伝導マイク４１が検出し、電気的な音信号に変換してスロットマシン１に出力する。したがって、他の遊技者による音声や周囲の環境音等の雑音はスロットマシン１に入力されず、ヘッドセット４０を装着している遊技者Ｐが発声する音声のみでスロットマシン１を操作することができる。これにより、スロットマシン１の音声入力操作において、このような雑音による誤作動を防止することができる。

【0061】

また、スロットマシン１が送出する、ゲームを演出するための効果音等を内蔵するスピーカ４２を介して遊技者Ｐの耳に直接供給する。したがって、遊技者Ｐは、周囲の環境音等に干渉されずゲームの演出効果音を楽しむことができる。

【0062】

次に、図５のブロック図を参照して、本スロットマシン１に設けられている制御システムについて説明する。

【0063】

主制御基板３００は、演算装置としてのＣＰＵと半導体メモリであるＲＯＭ、及びＲＡＭ等の電子部品が搭載されており、ＲＯＭに予め記憶されているシステムプログラムとスロットマシンゲーム用のプログラムに従ってＣＰＵが演算処理を行う。そして、主制御基板３００は、サブ制御基板２００及び回胴装置基板３１０等の各制御基板に所定の指令内容を含む制御信号を送出することで分散制御を行わせ、これによりスロットマシン１全体を統括して制御する。

【0064】

また、主制御基板３００には、中央表示基板２４０が接続されている。中央表示基板２４０には、メダル投入センサ２４８と、ベットボタン２４２，２４３，２４４と、スタートレバー２４５と、ストップボタン２４６ａ，２４６ｂ，２４６ｃ等のスイッチ及びセンサが接続され、これらスイッチ類から出力される出力信号を主制御基板３００へ転送する中継基板として機能する。

【0065】

また、中央表示基板２４０には、後述する音声認識部５０と操作モード切換回路４４３が接続されている。主制御基板３００は、操作モード切換回路４４３の短絡を検出すると、スロットマシン１本体に上述のヘッドセット４０が正しく接続された状態にあると判断し、音声認識部５０の動作を有効にする。これにより、ヘッドセット４０を装着する遊技者が発声するに応じてスロットマシン１にゲーム動作をさせる操作モード（ここでは、本操作モードを「音声操作モード」と呼んでいる）に切り換える。音声操作モード中は、スタートレバー２４５またはストップボタン２４６ａ，２４６ｂ，２４６ｃからの操作信号を無効に（受付けないように）してもよい。

【0066】

サブ制御基板２００は、主に演出を制御実行するための制御基板である。サブ制御基板２００には、演出用ランプ２１３，２２６ａ，２２６ｂ等の発光源である高輝度の発光ダイオードに対し点灯・点滅させる制御を行うための照明制御部２０１と、演出効果音を出力するスピーカ２５１ａ，２５１ｂ，２５２ａ，２５２ｂを鳴動制御するための音響制御部２０２と、ＶＤＰ（Video Display Processor）やフレームメモリまたはビデオＲＡＭ等を搭載し液晶表示装置２１１に演出映像を表示させる制御を行うための表示装置制御部２０３と、ドット表示装置２２４や遊技情報表示装置２２３等に遊技情報を表示させる制御を行う遊技情報表示制御部２０４とを備えている。

【0067】

サブ制御基板２００は、主制御基板３００から演出開始を示唆する制御信号が供給されると、これら演出用ランプ２１３，２２６ａ，２２６ｂ及びスピーカ２５１ａ，２５１ｂ，２５２ａ，２５２ｂ等を駆動し、また液晶表示装置２１１に演出映像を表示することによって、遊技者の視覚と聴覚に訴える演出を行う。

【0068】

回胴装置基板３１０には、リール３１１ａ，３１１ｂ，３１１ｃを回転駆動するためのステッピングモータ３１２ａ，３１２ｂ，３１２ｃが接続され、主制御基板３００からリール３１１ａ，３１１ｂ，３１１ｃを回転または停止させるべき旨の制御信号が供給されると、回胴装置基板３１０はステッピングモータ３１２ａ，３１２ｂ，３１２ｃへの駆動パルス信号の周期またはパルス幅を制御してリール３１１ａ，３１１ｂ，３１１ｃの回転動作を制御する。

【0069】

主制御基板３００に接続されている電源装置基板３３０は、主電源装置３３１で発生される各種電源電圧をホッパ装置３２１その他の各所に配電する配電回路が形成されており、かかる配電回路からスロットマシン１の動作に必要なシステム電源を供給する。また、電源装置基板３３０は、主制御基板３００からメダルの払い出しをすべき旨の指令を受けると、ホッパ装置３２１を制御して排出口２３２からのメダルの払い出しを行わせる。

【0070】

次に、音声認識部５０の構成を図６のブロック図及び図７の表を参照して説明する。なお、図面上で特には示していないが、音声認識部５０はＣＰＵ及びＤＳＰ（Digital Signal Processor）を備えており、これらが連携してヘッドセット４０の骨伝導マイク４１からの音信号を入力してデジタル演算処理するものである。特に、音声認識部５０の前段部分は、主にＤＳＰにより実行される信号処理ブロックであるデジタルサンプリング部５１と、音声抽出フィルタ５２と、スペクトル分析部５３とから構成されている。

【0071】

図６において、骨伝導マイク４１が出力する音信号は音声認識部５０に入力され、デジ

タルサンプリング部５１において所定周期のデジタルサンプル値信号に変換される。音声抽出フィルタ５２は、人間の音声の主要な周波数帯域のみを通過させるデジタルバンドパスフィルタとして形成され、音信号のサンプル値信号から人間の音声に係る音声信号を主に抽出する。そして、当該周波数帯域を通過する所定の信号レベル以上の信号のみを音声信号として次のスペクトル分析部５３へ出力する。

【0072】

スペクトル分析部５３では、上述の音声信号から音素（母音、子音、撥音）の情報を含んでいる音調フィルタ伝達関数を推定する。ここで、音声信号は人間の声帯から発せられた音源信号と、音源信号を口や鼻、舌によって音声信号に変調する特性である、いわゆる音調フィルタ伝達関数とが畳み込まれた信号としてモデル化することができる。つまり、スペクトル分析部５３での信号処理は、音声信号から音源信号と音調フィルタ伝達関数とを分離し、分離した音調フィルタ伝達関数からそれに含まれる音素情報を推定することを目的とするものである。かかる信号処理の具体的な手法としてはケプストラム（Cepstrum）分析法等が代表的な手法として用いられ、スペクトル分析部５３では、数十msec程度の定常とみなし得る音声区間（フレーム）毎の音声信号から上記音調フィルタ伝達関数のパワースペクトルを求めている。

【0073】

次に、ワード変換部５４において、前段で抽出された音素情報を含むパワースペクトルの形状（音声信号の動的な特徴）に対応する音素を推定する。すなわち、求められたフレームのスペクトルと予め分かっている音素スペクトルとの相関をとることで音声信号に含まれていた音素を推定する。そして、求めた音素を時系列に並べることにより一つの単語（ワード）に変換し、これを遊技者が発声した単語として認識する。

【0074】

ワード登録管理部５５には、スロットマシン１の音声操作モードにおける遊技動作と、その操作を行うための登録された操作命令語とが対応づけられた図７に示される操作命令語管理テーブル５５１がデータベース化されて備えられている。図７に示されるように、操作を行うための操作命令語は、例えば“ひだり”、“みぎ”等、簡単な単語から形成されている。

【0075】

例えば、操作命令語管理テーブル５５１のＮｏ．１のカラムでは、“ひだり”という操作命令語に対して「第１リール（３１１ａ）を停止する」という遊技動作が対応している。また、Ｎｏ．２のカラムでは、“なか”という操作命令語に対して「第２リール（３１１ｂ）を停止する」という遊技動作が対応している。このように、ワード登録管理部５５において予め登録されている操作命令語には、スロットマシン１に対して行う何らかの遊技動作が対応づけられている。

【0076】

また、一つの遊技動作に複数の操作命令語が対応づけられている。例えば、「第１リール（３１１ａ）を停止する」という遊技動作に対応して、Ｎｏ．１カラムの“ひだり”と、Ｎｏ．１１カラムの“ひ”と、Ｎｏ．２１カラムの“いち”と、Ｎｏ．３１カラムの“ワン”等が、操作命令語として予め登録されている。

【0077】

また、操作命令語管理テーブル５５１のＮｏ．１０１カラム及びＮｏ．１０２カラムのように、停止させるリールを特定しない操作命令語（例えば“とまれ”または“ストップ”等）が登録されている。

【0078】

また、スロットマシン１への操作の内容は、リールを停止する上述の操作に限られず、例えば、操作命令語管理テーブル５５１のＮｏ．２０１及びＮｏ．２０２カラムにあるように、リール３１１ａ，３１１ｂ，３１１ｃを一斉に回転させるための操作命令語（“まわれ”、“スタート”等）が本遊技動作に対応づけられて登録されている。

【0079】

操作命令語管理テーブル５５１のカラムを追加することで、遊技者が所望する任意の操作命令語の登録が行われる。すなわち、新たに操作命令語を登録するには、例えば手入力されるスイッチ等の操作手段を操作しながら、その操作命令語を骨伝導マイク４１に入力することで自動的に登録される。より具体的には、第１リール３１１ａを停止させるための操作命令語として“ピン”という単語を登録する場合には、ストップボタン２４６ａを長押ししながら“ピン”と発声し骨伝導マイク４１に入力する。音声認識部５０では、音声認識部５０のワード変換部５４がこの単語（“ピン”）を認識し、この単語情報をワード登録管理部５５へ転送し、ワード登録管理部５５がこの単語を当該操作による遊技動作のための操作命令語として対応づけ、操作命令語管理テーブル５５１のカラムに追加する。このように、新たな操作命令語の登録を簡易的に行うことができる。また、より簡略化した短い任意の単語を操作命令語として登録することで、簡略化した単語を使ったスムーズな音声操作を行うことができる。

【0080】

また、ある操作命令語の登録を無効にする場合には、操作命令語管理テーブル５５１から当該操作命令語が登録されたカラムを削除すればよい。

【0081】

なお、操作命令語管理テーブル５５１における遊技動作は、実際にはその遊技動作に結びつく操作命令コードに符号化されて当該テーブルに記述されていることは言うまでもない。

【0082】

図６に戻り、ワード比較照合部５６においては、ワード変換部５４までの処理で音声認識した単語がワード登録管理部５５で管理されている操作命令語管理テーブル５５１に記述された操作命令語と一致するかどうか比較照合する。ワード比較照合部５６の一つの動作例は下記のようになる。

【0083】

ワード比較照合部５６はワード登録管理部５５の操作命令語管理テーブル５５１を参照し、もし音声認識した単語と一致する操作命令語があれば、その操作命令語に対応付けられた遊技動作を実行させるためのコマンド信号をスイッチ信号出力インタフェース５７に送出する。

【0084】

スイッチ信号出力インタフェース５７は、ワード比較照合部５６からのコマンド信号が第１リール３１１ａの回転停止を行うためのものである場合、出力端子５８ａを介して電気的なスイッチ信号（ＯＮ信号）を上述の中央表示基板２４０へ出力する。同様に、第２リール３１１ｂの回転停止を行うためのコマンド信号を受けると出力端子５８ｂを介して、第３リール３１１ｃの回転停止を行うためコマンド信号を受けると出力端子５８ｃを介してスイッチ信号を出力する。

【0085】

また、上述のコマンド信号の指令がリールの回転を開始させるものであるとき、スイッチ信号出力インタフェース５７は、出力端子５９を介して全てのリール３１１ａ，３１１ｂ，３１１ｃを回転させるためのスイッチ信号を出力する。

【0086】

例えば、骨伝導マイク４１に入力された音信号から“スタート”という単語がワード変換部５４までの信号処理で音声認識された場合、“スタート”という単語は全てのリール３１１ａ，３１１ｂ，３１１ｃの回転を開始させる遊技動作に対応づけられた操作命令語であるとワード比較照合部５６が判定すると、その遊技動作を示唆するコマンド信号をスイッチ信号出力インタフェース５７に出力する。スイッチ信号出力インタフェース５７はこれを受け、リール３１１ａ，３１１ｂ，３１１ｃの回転開始を行うための電気的なスイッチ信号（ＯＮ信号）を出力端子５９を介して中央表示基板２４０へ出力するように構成されている。

【0087】

また、骨伝導マイク４１に入力された音信号から“ひだり”という単語が音声認識された場合、“ひだり”という単語は第１リール３１１ａを停止する遊技動作に対応づけられた操作命令語であるとワード比較照合部５６が判定すると、ワード比較照合部５６はその遊技動作を示唆するコマンド信号をスイッチ信号出力インタフェース５７に出力する。そして、スイッチ信号出力インタフェース５７は、第１リール３１１ａの回転停止を行うための電気的なスイッチ信号（ＯＮ信号）を出力端子５８ａを介して中央表示基板２４０へ出力する。

【0088】

このような音声認識部５０における信号処理により、遊技者が骨伝導マイク４１に入力した音声に応じて、リール３１１ａ，３１１ｂ，３１１ｃの回転を開始させたり、停止させたりする等の操作制御を実現することができる。

【0089】

次に、かかる構成の本スロットマシン１の動作例を図８のフローチャートを参照して説明する。なお、ここでは、ヘッドセット４０を装着した遊技者が骨伝導マイク４１を介して音声を入力することによりスロットマシン１を操作する、音声操作モードにおける動作を説明する。

【0090】

本スロットマシン１は、遊技者によってメダル投入部２４１にメダルが投入され主制御基板３００がメダル投入センサ２４８からのメダル投入の検出信号を受けたとき、または何れかのベットボタン２４２，２４３，２４４が押圧操作されクレジット（貯留）されているメダルからゲームに賭けるメダルの枚数が提示されたとき、図８に示される当該ゲームが開始される。

【0091】

ステップＳ１０１において、遊技者が骨伝導マイク４１に単語を入力し、音声認識部５０において音声認識した当該単語の情報がリール３１１ａ，３１１ｂ，３１１ｃの回転を開始させるための登録されている所定の単語、すなわち、ここでいう操作命令語（例えば、“まわれ”または“スタート”等）であると判定すると、ステップＳ１０２において、全てのリール３１１ａ，３１１ｂ，３１１ｃの回転を開始する。

【0092】

リール３１１ａ，３１１ｂ，３１１ｃの回転速度が一定（例えば８０rpm）になると、各リールに対する停止操作が準備状態となり、ストップボタン２４６ａ，２４６ｂ，２４６ｃに内蔵されている青の発光ダイオードを点灯する。そして、遊技者は、回転しているリールに対し、図柄が揃うようにタイミングを見計らいながら各リール３１１ａ，３１１ｂ，３１１ｃを停止させる音声による操作を行う。

【0093】

すなわち、ステップＳ１０３において、遊技者が骨伝導マイク４１に単語を入力し、音声認識した単語情報が何れかのリールを停止させるための登録されている操作命令語であると判定すると、ステップＳ１０４に移行する。ステップＳ１０４において、音声による当該操作が有効に受付けられたことを遊技者に報知する。すなわち、当該リール（音声により停止操作の対象としたリール）に対応するストップボタンによる発光表示を青から赤に変更して表示する。同時に、確認音等を鳴らしてもよい。

【0094】

ステップＳ１０５において、操作を受付けた当該リールを制動して停止させる。

【0095】

遊技者の音声入力操作に応じて各リール３１１ａ，３１１ｂ，３１１ｃの回転を停止させるステップＳ１０３～ステップＳ１０５までの処理は、これら全てのリールに対して行われる。

【0096】

ステップＳ１０６において、全てのリール３１１ａ，３１１ｂ，３１１ｃが停止したと判定すると、ステップＳ１０７へ移行する。ステップＳ１０７では、主制御基板３００は

、停止した各リール３１１ａ，３１１ｂ，３１１ｃが表示する図柄を調べ、図柄が揃っていなければゲーム終了となり、図柄が入賞に係る組合せに揃ったと判定すると、ステップＳ１０８へ進み、当該入賞の種類に応じた枚数のメダルを遊技者へ払い出す。

【0097】

かかる構成の本スロットマシン１によれば、接続可能なヘッドセット４０が設けられ、骨伝導マイク４１に入力された操作命令語に応じてリール３１１ａ，３１１ｂ，３１１ｃの回転開始、及び停止の操作を行うことができる。このような音声を介して行われる操作機能により、ゲームの操作における遊技者の手への負担を軽減することができる。したがって、長時間にわたってゲームを行ったとしても遊技者の手に疲労が蓄積することはなくなる。

【0098】

また、ヘッドセット４０に設けられた骨伝導マイク４１を介して音声入力操作を行うので、他の遊技者による音声や周囲の環境音等の雑音がスロットマシン１に入力されず、ヘッドセット４０を装着している遊技者が発声する音声のみでスロットマシン１を操作することができる。これにより、スロットマシン１の音声入力操作において、雑音による誤作動を防止することができる。

【図面の簡単な説明】

【0099】

【図1】従来技術による、スロットマシンの外観構造を表した正面図である。

【図2】本発明の一実施態様による、スロットマシンの外観構造を表した正面図である。

【図3】本発明の一実施態様による、スロットマシンのコネクタソケットに接続されたヘッドセットの構成を表した側面図である。

【図4】本発明の一実施態様による、骨伝導マイクが備えられたヘッドセットの耳あて部の構造を表した断面図である。

【図5】本発明の一実施態様による、スロットマシンの制御システムを表したブロック図である。

【図6】本発明の一実施態様による、音声認識部の構成を表したブロック図である。

【図7】本発明の一実施態様による、音声認識部に備えられる操作命令語管理テーブルを表した図である。

【図8】本発明の一実施態様による、スロットマシンの動作を表したフローチャートである。

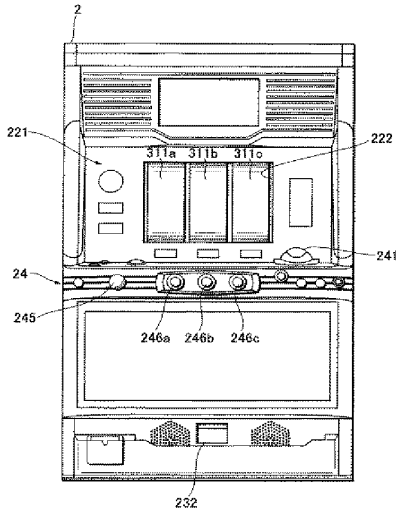
【符号の説明】

【0100】

１…スロットマシン、　２…フロントドア、　３…筐体、
２００…サブ制御基板、
２３２…排出口、　２３３…受皿、
２４…操作卓、　２４１…メダル投入部、
２４２，２４３，２４４…ベットボタン、
２４５…スタートレバー、
２４６ａ，２４６ｂ，２４６ｃ…ストップボタン、
２４７…コネクタソケット、
２４０…中央表示基板、３００…主制御基板、
３１…回胴装置、　３１１ａ，３１１ｂ，３１１ｃ…リール、
４０…ヘッドセット、４０１…耳あて部、　４０６…ヘッドバンド
４１…骨伝導マイク、　４２…スピーカ、
４３…コード、　４４…コネクタプラグ
５０…音声認識部、
５１…デジタルサンプリング部、　５２…音声抽出フィルタ、
５３…スペクトル分析部、　５４…ワード変換部、
５５…ワード登録管理部、　５５１…操作命令語管理テーブル、

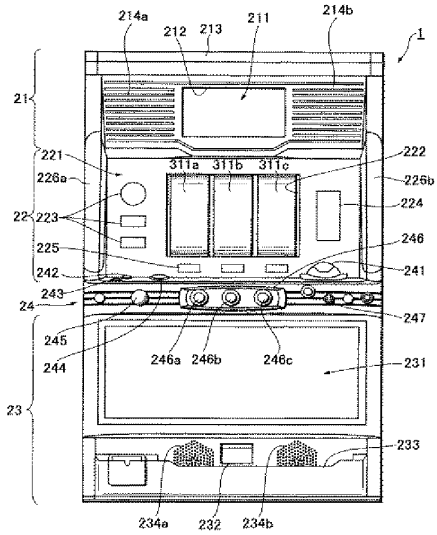
５６…ワード比較照合部、　５７…スイッチ信号出力インタフェース

【図1】

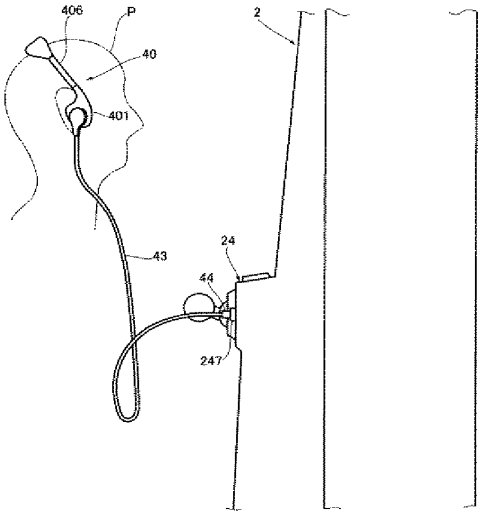


従来技術

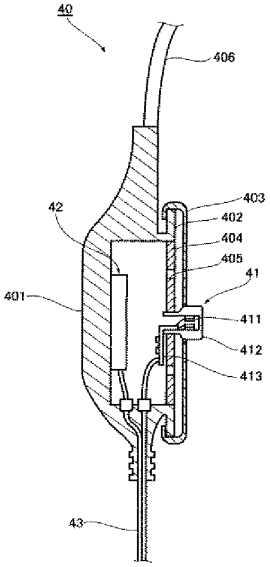
【図2】



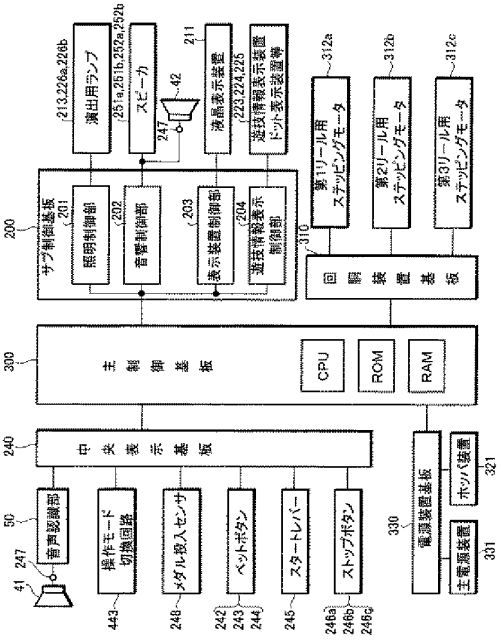
【図3】


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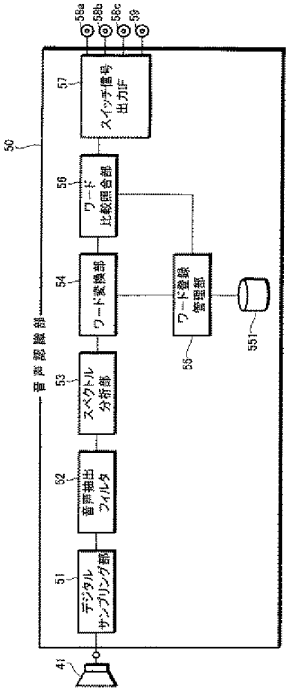
【図4】


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【図5】


演出用ランプ
スピーカ
液晶表示装置
遊技情報表示装置
ドット表示装置等
第1リール用ステッピングモータ
第2リール用ステッピングモータ
第3リール用ステッピングモータ
サブ制御基板
照明制御部
音響制御部
表示装置制御部
遊技情報表示制御部
回胴装置基板
主制御基板
CPU
ROM
RAM
中央表示基板
音声駆動部
操作モード切換回路
メダル投入センサ
ベットボタン
スタートレバー
ストップボタン
電源装置基板
ホッパ装置
主電源装置

【図6】

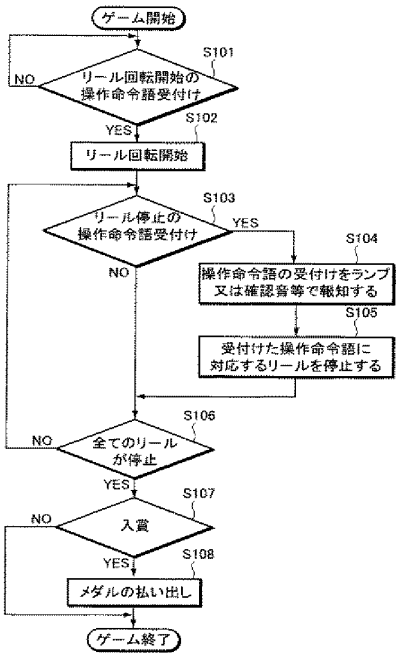

音声部駆動部
スイッチ信号出力IF
ワード比較照合部
ワード登録機部
ワード登録管理部
スペクトル分析部
音声抽出フィルタ
デジタルサンプリング部

【図7】

表　操作命令語管理テーブル(551)

No.	登録された操作命令語	遊技動作(操作内容)
1	"ひだり"	第1リール(311a)を停止する
2	"なか"	第2リール(311b)を停止する
3	"みぎ"	第3リール(311c)を停止する
：	：	：
11	"ひ"	第1リール(311a)を停止する
12	"な"	第2リール(311b)を停止する
13	"み"	第3リール(311c)を停止する
：	：	：
21	"いち"	第1リール(311a)を停止する
22	"に"	第2リール(311b)を停止する
23	"さん"	第3リール(311c)を停止する
：	：	：
31	"ワン"	第1リール(311a)を停止する
32	"ツー"	第2リール(311b)を停止する
33	"スリー"	第3リール(311c)を停止する
：	：	：
101	"とまれ"	リールを停止する
102	"ストップ"	リールを停止する
：	：	：
201	"まわれ"	リールを回転させる
202	"スタート"	リールを回転させる
：	：	：

【図8】



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(81) 指定国 (国内): US.

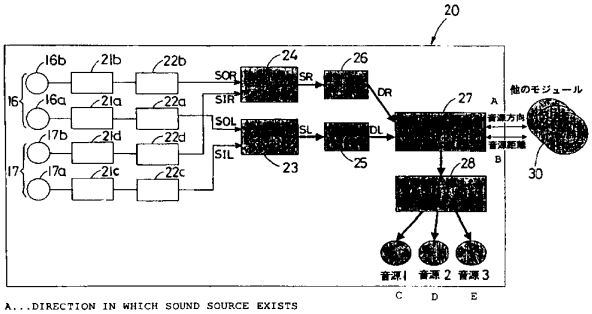
(84) 指定国 (広域): ヨーロッパ特許 (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE, TR).

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２文字コード及び他の略語については、定期発行される各PCTガゼットの巻頭に掲載されている「コードと略語のガイダンスノート」を参照。

(54) Title: ROBOT HEARING SYSTEM

(54) 発明の名称: ロボット聴覚システム



A...DIRECTION IN WHICH SOUND SOURCE EXISTS
B...DISTANCE TO SOUND SOURCE
30...OTHER MODULES
C...SOUND SOURCE 1
D...SOUND SOURCE 2
E...SOUND SOURCE 3

(57) Abstract: A hearing system (10) of a robot, especially the one of human or animal type, capable of collecting sound from an external target and actively hearing it without being influenced by the noise produced by, e.g. a drive mechanism in the robot. The system comprises a pair of external microphones (16) provided at the ear parts outside the sound-insulating shell (13), a pair of internal microphones (17) for collecting noise provided inside the shell, noise removers (23, 24) for removing noise signals from an inside noise source from the acoustic signals from the external microphones, pitch extracting sections (25, 26) for extracting acoustic data (DL, DR) about the time, frequency and power from the left and right acoustic signals from the noise removers by frequency analysis, a left/right channel relating section (27) for determining the direction from which sound comes on the basis of the harmonic structure involving the pitches of the left/right acoustic data extracted by the pitch extracting section, and a sound source separating section (28) for separating acoustic data into acoustic data sets of respective sound sources according to the information on the sound direction determination.

[続葉有]

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(57) 要約:

本発明は駆動機構等のロボット内部で発生するノイズに影響されることなく、外部の目標からの音を集音して能動知覚を行なうことができるようにしたロボット、特に人型または動物型ロボットの聴覚システム（１０）を提供するもので、防音性の外装（１３）の外側の両耳位置に設けた一対の外部マイク（１６）と、外装の内側に設けたノイズ集音用の一対の内部マイク（１７）と、外部マイクからの音響信号から内部のノイズ発生源からのノイズ信号をキャンセルするノイズ除去部（２３，２４）と、ノイズ除去部からの左右の音響信号からそれぞれ周波数解析により時間，周波数及びパワーに関する音響データＤＬ，ＤＲを取り出すピッチ抽出部（２５，２６）と、ピッチ抽出部で取り出した左右の音響データのピッチを伴う調波構造から音の方向付けを行なう左右チャンネル対応部（２７）と、音の方向付け情報に基づいて、音響データから音源毎の音響データに分離する音源分離部（２８）とを具備するものである。

明　　細　　書

ロボット聴覚システム

技術分野

本発明はロボット、特に人型または動物型ロボットにおける聴覚システムに関する。

背景技術

近年、このような人型または動物型ロボットにおいては、視覚，聴覚の能動知覚が注目されてきている。

能動知覚とは、ロボット視覚やロボット聴覚等の知覚を担当する知覚装置を、知覚すべき目標に追従するように、これらの知覚装置を支持する、例えば頭部を駆動機構により姿勢制御するものである。

ここで、能動視覚に関しては、少なくとも知覚装置であるカメラが、駆動機構による姿勢制御によってカメラの光軸方向が目標に向かって保持され、目標に対して自動的にフォーカシングやズームイン，ズームアウト等が行なわれることにより、カメラが目標を撮像するようになっている。このような能動視覚については、種々の研究が行なわれている。

これに対して、能動聴覚に関しては、少なくとも知覚装置であるマイクが、駆動機構による姿勢制御によって、その指向性が目標に向かって保持され、目標からの音がマイクによって集音される。この際、能動聴覚の不利な点として、駆動機構が作動している間はマイクが駆動機構の作動音を拾ってしまうため、目標からの音に比較的大きなノイズが混入してしまい、目標からの音を認識できなくなってしまうことがある。

しかしながら、駆動機構を停止させた状態での聴覚研究では、特に目標が移動しているような場合、目標の移動に追従しながらの所謂能動聴覚を行なうことが困難であった。

さらに、上述した駆動機構だけでなく、ロボット内部で発生する各種動作音及

び定常的に発生するノイズも聴覚装置としてのマイクが集音してしまうので、同様に完全な能動聴覚を得ることが困難であった。

本発明の目的は、このような従来の技術における欠点を解決するものであり、駆動機構等のロボット内部で発生するノイズに影響されることなく、外部の目標からの音を集音して能動知覚を行なうことができるようにしたロボット、特に人型または動物型ロボットの聴覚システムを提供することにある。

発明の開示

この目的を達成するために、本発明によるロボット聴覚システムは、内部にノイズ発生源を備えたロボットにおいて、少なくともロボットの一部を覆う防音性の外装と、上記外装の外側に設けられ主として外部の音を集音する少なくとも一対の外部マイクと、上記外装の内側に設けられ主として内部のノイズ発生源からのノイズを集音する少なくとも一対の内部マイクと、上記外部マイク及び内部マイクからの左右の信号に基づいてそれぞれ外部マイクからの音響信号から内部のノイズ発生源からのノイズ信号をキャンセルするノイズ除去部と、上記ノイズ除去部からの左右の音響信号からそれぞれ周波数解析を行なって、時間，周波数及びパワーに関する音響データ及びピッチ情報を取り出すピッチ抽出部と、上記ピッチ抽出部で取り出された左右の音響データ及びピッチ情報に基づいて当該音響データが示すピッチを伴う調波構造から音の方向付けを行なう左右チャンネル対応部と、上記左右チャンネル対応部で取り出された音の方向付け情報に基づいて上記音響データから音源毎の音響データに分離する音源分離部と、を含んでいることを特徴とする。

また、上記目的を達成するために、本発明による人型または動物型ロボットの聴覚システムは、内部に駆動機構等のノイズ発生源を備えた人型または動物型のロボットにおいて、少なくともロボットの頭部を覆う防音性の外装と、上記外装の外側にて両側の耳に対応する両耳位置に設けられ主として外部の音を集音する少なくとも一対の外部マイクと、上記外装の内側に設けられた主として内部のノイズ発生源からのノイズを集音する少なくとも一対の内部マイクと、上記外部マイク及び内部マイクからの信号に基づいて外部マイクからの音響信号から内部の

ノイズ発生源からのノイズ信号をキャンセルするノイズ除去部と、上記ノイズ除去部からの左右の音響信号からそれぞれ周波数解析を行なって時間，周波数及びパワーに関する音響データ及びピッチ情報を取り出すピッチ抽出部と、上記ピッチ抽出部で取り出された左右の音響データピッチ情報に基づいて当該音響データのピッチを伴う調波構造から音の方向付けを行なう左右チャンネル対応部と、上記左右チャンネル対応部で取り出された音の方向付け情報に基づいて上記音響データから音源毎の音響データに分離する音源分離部と、を含んでいることを特徴としている。

上記構成によれば、外部マイクが主として外部の目標からの音を集音し、内部マイクが主としてロボット内部の駆動機構等のノイズ発生源からのノイズを集音する。その際、外部マイクが集音した音響信号には、ロボット内部のノイズ発生源からのノイズ信号が混入しているが、この混入したノイズ信号は、ノイズ除去部における演算処理によって内部マイクが集音したノイズ信号によりキャンセルされることにより、著しく低減される。

そして、ノイズがキャンセルされた音響信号から、ピッチ抽出部により周波数解析によって音響信号を取り出して、この音響信号から左右チャンネル対応部により音の方向付けを行なった後、この音の方向付けに基づいて、音源分離部によって各音源毎の音響データに分離する。

したがって、ノイズ除去部での演算処理によって、ロボット内部の駆動機構等のノイズ発生源からのノイズが容易に著しく低減されることにより、外部マイクからの音響信号のＳ／Ｎ比が大幅に改善されるので、音源毎の音響データの分離をより一層良好に行なうことができる。

本発明によるロボット聴覚システムは、好ましくは、さらに、ロボットが視覚，触覚等の知覚システムを備えており、上記左右チャンネル対応部がこれらの知覚システムからの情報を参照して音の方向付けを行なう。

この構成によれば、例えば視覚装置からの目標に関する視覚情報に基づいて左右チャンネル対応部がより明確な方向付けを行なうことができる。

本発明によるロボット聴覚システムは、好ましくは、上記左右チャンネル対応部が上記知覚システムに対して音の方向付けに関する情報を出力する。

この構成によれば、例えば視覚装置に対して目標に関する聴覚による方向付けの情報を出力するので、視覚装置がより一層正確な方向付けを行なうことができる。

さらに、本発明によるロボット聴覚システムは、好ましくは、上記内部マイクが、外装の内側において少なくとも横方向に移動可能に配設されていることを特徴とする。

この構成によれば、ロボット内部のノイズ発生源からのノイズをより良好に集音するように内部マイクを横方向、前後方向及び／又は上下方向に移動させて、各内部マイクと対となるべき対応する外部マイクとの距離を適宜に調整することができ、外部マイクが集音する音響信号から最も最適な状態でノイズをキャンセルすることができる。

本発明によるロボット聴覚システムは、好ましくは、上記内部マイクが、外装の内側にて横方向に延びる支持部材に対して摺動可能に支持されている。

本発明によるロボット聴覚システムは、好ましくは、上記支持部材が横方向に延びるスロットを備えており、上記内部マイクが上記支持部材のスロットに沿って摺動可能に支持された可動部材に固定されている。

本発明によるロボット聴覚システムは、好ましくは、上記可動部材が、スロットに挿通された固定ネジにより支持部材に対して螺着される。

本発明によるロボット聴覚システムは、好ましくは、上記内部マイクがさらに前後方向に移動可能に配設されている。

本発明によるロボット聴覚システムは、好ましくは、上記内部マイクが外装の内側にて横方向に延びる第一の支持部材に対して摺動可能に支持されていて、この第一の支持部材が外装の内側にて前後方向に延びる第二の支持部材に対して摺動可能に支持されている。

本発明によるロボット聴覚システムは、好ましくは、上記内部マイクがさらに上下方向に移動可能に配設されている。

図面の簡単な説明

本発明は、以下の詳細な説明及び本発明の実施例を示す添付図面に基づいて、

より良く理解されるものとなろう。なお、添付図面に示す実施例は、本発明を特定することを意図するものではなく、単に説明及び理解を容易にするためのものである。

図中、

図１は、本発明によるロボット聴覚装置の第一の実施形態を組み込んだ人型ロボットの外観を示す正面図である。

図２は、図１の人型ロボットの側面図である。

図３は、図１の人型ロボットにおける頭部の構成を示す概略拡大図である。

図４は、図１の人型ロボットにおけるロボット聴覚システムの電気的構成を示すブロック図である。

図５は、図１の人型ロボットにおける内部マイクの取付構造を示す概略正面図である。

図６は、図５の内部マイクの取付構造の要部を示す拡大正面図である。

図７は、図５の内部マイクの取付構造の要部を示す拡大側面図である。

図８は、図５の内部マイクの取付構造の要部を示す拡大斜視図である。

図９は、図４のロボット聴覚システムにおけるピッチ抽出部により取り出された音響データ、ピッチ情報の一例を示すグラフである。

図１０は、図５の内部マイクの取付構造の変形例の要部を示す拡大斜視図である。

発明を実施するための最良の形態

以下、本発明のロボット聴覚システムの最良の第一の実施例を図面を参照して詳細に説明する。

図１から図４は本発明によるロボット聴覚システムの一実施形態を備えた実験用の人型ロボットの全体構成を示している。

図１において、人型ロボット１０は、４ＤＯＦ（自由度）のロボットとして構成されている。人型ロボット１０は、ベース１１と、ベース１１上にて一軸（垂直軸）周りに回動可能に支持された胴体部１２と、胴体部１２上にて三軸方向（垂直軸，左右方向の水平軸及び前後方向の水平軸）の周りに揺動可能に支持され

た頭部１３と、を含んでいる。

上記ベース１１は、固定配置されていてもよく、ロボットの脚部として動作可能としてもよい。また、ベース１１は移動可能な台車等の上に載置されていてもよい。

上記胴体部１２は、ベース１１に対して垂直軸の周りに、図１にて矢印Ａで示すように回動可能に支持されており、図示しない駆動手段によって回転駆動される。さらに、胴体部１２は、全体が防音性の外装によって覆われている。

上記頭部１３は胴体部１２に対して連結部材１３ａを介して支持されており、この連結部材１３ａに対して前後方向の水平軸の周りに、図１にて矢印Ｂで示すように揺動可能に、また左右方向の水平軸の周りに、図２にて矢印Ｃで示すように揺動可能に支持されている。この上記連結部材１３ａは、胴体部１２に対してさらに前後方向の水平軸の周りに、図１にて矢印Ｄで示すように揺動可能に支持されている。

そして、上記矢印Ａ，Ｂ，Ｃ，Ｄ方向の回転駆動は、それぞれ図示しない駆動手段によって行なわれる。

ここで、上記頭部１３は、図３に示すように全体が防音性の外装１４により覆われている。さらに、上記頭部１３は、前側にロボット視覚を担当する視覚装置としてのカメラ１５を、また両側にロボット聴覚を担当する聴覚装置としての一対の外部マイク１６（１６ａ，１６ｂ）を備えている。

さらに、上記頭部１３は、図３に示すように、外装１４の内側にて左右に離隔して配設された一対の内部マイク１７（１７ａ，１７ｂ）を備えている。

上記外装１４は、例えばウレタン樹脂等の吸音性の合成樹脂から構成されている。上記外装１４は、頭部１３の内部をほぼ完全に密閉することにより頭部１３の内部の遮音を行なうように構成されている。なお、胴体部１２の外装も、同様にして吸音性の合成樹脂から構成されている。

上記カメラ１５は公知の構成であって、例えば所謂パン，チルト，ズームの３ＤＯＦ（自由度）を有する市販のカメラを使用することができる。

上記外部マイク１６は、それぞれ頭部１３の側面において、前方に向かって指向性を有するように取り付けられている。

ここで、外部マイク１６の左右の各外部マイク１６ａ，１６ｂは、それぞれ図１及び図２に示すように、外装１４の両側にて前方に向いた段部１４ａ，１４ｂにて内側に取り付けられ、段部１４ａ，１４ｂに設けられた貫通穴を通して前方の音を集音すると共に、外装１４の内部の音を拾わないように適宜の手段により遮音されている。

これにより外部マイク１６ａ，１６ｂは、所謂バイノーラルマイクとして構成されている。なお、外部マイク１６ａ，１６ｂの取付位置の近傍において、外装１４は人間の外耳形状に形成されていてもよい。

上記内部マイク１７は、それぞれ外装１４の内側において、上述した各外部マイク１６ａ，１６ｂの近傍に、図示の場合には、前記カメラ１５の両端付近の上方にて、後述するように横方向に移動可能に配設されている。

なお、上記内部マイク１７は、外装１４の内側にて任意の位置に備えられていてもよい。

図４は上記外部マイク１６及び内部マイク１７を含む音響処理のための聴覚システムの電気的構成を示している。

図４において、聴覚システム２０は、各外部マイク１６ａ，１６ｂ及び内部マイク１７ａ，１７ｂからの音響信号をそれぞれ増幅するアンプ２１ａ，２１ｂ，２１ｃ，２１ｄと、これらの増幅信号をＡＤ変換によりデジタル音響信号ＳＯＬ，ＳＯＲ，ＳＩＬ，ＳＩＲに変換するＡＤコンバータ２２ａ，２２ｂ，２２ｃ，２２ｄと、これらのデジタル音響信号が入力される左右のノイズ除去回路２３，２４と、ノイズ除去回路２３，２４からのデジタル音響信号ＳＲ，ＳＬが入力されるピッチ抽出部２５，２６と、ピッチ抽出部２５，２６からの音響データが入力される左右チャンネル対応部２７と、左右チャンネル対応部２７からのデータが入力される音源分離部２８と、から構成されている。

上記ＡＤコンバータ２２ａ乃至２２ｄは、例えば量子化ビット数１６または２４で、４８ｋＨｚサンプリングした信号を取り出すように構成されている。

そして、左側の外部マイク１６ａからのデジタル音響信号ＳＯＬ及び左側の内部マイク１７ａからのデジタル音響信号ＳＩＬは、ノイズ除去回路２３に入力される。また、右側の外部マイク１６ｂからのデジタル音響信号ＳＯＲ及び左側の

内部マイク１７ｂからのデジタル音響信号ＳＩＲは、ノイズ除去回路２４に入力される。

これらのノイズ除去回路２３，２４は同一の構成であり、それぞれ外部マイク１６からの音響信号から内部マイク１７のノイズ信号によりノイズキャンセルするように構成されている。

即ち、ノイズ除去回路２３は、外部マイク１６ａからのデジタル音響信号ＳＯＬを、内部マイク１７ａが集音したロボット内部のノイズ発生源からのノイズ信号ＳＩＬに基づいて、例えば最も簡便には、外部マイク１６ａからのデジタル音響信号ＳＯＬから、内部マイク１７ａからの音響信号ＳＩＬを減算する等の適宜の演算処理によるノイズキャンセル処理によって、外部マイク１６ａからの音響信号ＳＯＬに混入したロボット内部の各駆動機構等のノイズ発生源からのノイズを除去して、左の音響信号ＳＬを生成する。

また、ノイズ除去回路２４は、外部マイク１６ｂからのデジタル音響信号ＳＯＲを、内部マイク１７ｂが集音したロボット内部のノイズ発生源からのノイズ信号ＳＩＲに基づいて、例えば最も簡便には、外部マイク１６ｂからのデジタル音響信号ＳＯＲから、内部マイク１７ｂからの音響信号ＳＩＲを減算する等の適宜の演算処理によるノイズキャンセル処理によって、外部マイク１６ｂからの音響信号ＳＩＲに混入したロボット内部の各駆動機構等のノイズ発生源からのノイズを除去して、右の音響信号ＳＲを生成する。

上記ピッチ抽出部２５，２６は同一の構成であり、ノイズ除去回路２３，２４からの左右の音響信号ＳＬ，ＳＲをそれぞれ周波数解析して、時間，周波数及びパワーの三軸から成る音響データ、並びにピッチ情報を取り出すように構成されている。即ち、ピッチ抽出部２５は、ノイズ除去回路２３からの左の音響信号ＳＬを周波数解析することにより、時間とパワーから成る二軸の音響信号ＳＬから時間，周波数及びパワーの三軸から成るスペクトログラムと呼ばれる左の音響データＤＬを取り出す。

また、ピッチ抽出部２６も同様にして、ノイズ除去回路２４からの右の音響信号ＳＲを周波数解析することにより、時間とパワーから成る二軸の音響信号ＳＲから時間，周波数及びパワーの三軸から成る右の音響データＤＲ、ピッチ情報を

取り出す。

ここで、上記周波数解析は、例えば２０m秒の窓長で、７．５m秒づつずらしながら、ＦＦＴ（高速フーリエ変換）を行なうことにより行なわれる。

なお、このような周波数解析は、ＦＦＴに限らず一般的な種々の方法で行なわれるようにしてもよい。

このようにして得られた音響データＤＬにおいては、音声や音楽における各音はスペクトログラム上にてピークの時間方向への連なりとして示されると共に、一般的に調波構造を有しており、整数倍の周波数値に規則的にピークを有するようになっている。

上記左右チャンネル対応部２７は、左右のピッチ抽出部２５，２６からの音響データＤＬ，ＤＲにてそれぞれピークにより示される調波構造のピッチ情報に関してその位相差，時間差等に基づいて、同じ音から派生するピッチを左右チャンネルで対応付けることにより音の方向付けを行なう。

図示の場合、左右チャンネル対応部２７は、聴覚システム２０以外にロボット１０に備えられた図示しない他の知覚システム３０から目標に関する情報、具体的には例えば視覚システムによる目標の位置，方向，形状，移動の有無等に関する情報や、触覚システムによる目標の柔軟度，振動の有無，触感等の情報が入力されることにより、上述した音源からの音の方向付けがより一層明確に行なわれるように構成されている。

上記音源分離部２８は、公知の構成により、左右チャンネル対応部２７からの方向付けの情報及び音響データＤＬ，ＤＲが入力されることにより、例えばディクションパスフィルタを使用する等の手法によって、方向付けの情報に基づいて音響データＤＬ，ＤＲから音源を同定すると共に、各音源毎に音響データを分離するように構成されている。

ここで、上記内部マイク１７は、具体的には、図５～図８に示すように、カメラ１５に対して取り付けられている。

先ず、図５において、各内部マイク１７ａ，１７ｂは、それぞれ可動部材としての取付金具４０，４１に対して固定されている。

この取付金具４０，４１は、それぞれ上方が開放したほぼＵ字形状に形成され

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ており、上端が締付ネジ４０ａ，４１ａで互いに近接されることにより内部マイク１７ａ，１７ｂを挟持して固定保持する。

また、上記取付金具４０，４１は、それぞれ支持部材４２，４３に対して後述するように横方向に摺動可能に取り付けられている。

上記支持部材４２，４３は、それぞれカメラ１５の上面に対してブラケット４４により固定されている。なお、図示の場合、支持部材４２，４３は、ブラケット４４と一体に形成されていると共に、ブラケット４４はカメラ１５が取り付けられる取付台１５ａに対して螺着されている。

ここで、各支持部材４２，４３は、それぞれ横方向に延びるスロット４２ａ，４３ａを備えている。

そして、上記取付金具４０，４１はそれぞれこれらのスロット４２ａ，４３ａに挿通された固定ネジ４５，４６を受容するネジ孔（図示せず）を有しており、これらの固定ネジ４５，４６が締め付けられることで支持部材４２，４３に対して固定保持される。

本発明実施形態による人型ロボット１０は以上のように構成されており、外部マイク１６ａ，１６ｂが集音しようとする目標からの音は、以下のようにして集音され、ノイズキャンセルされることにより音源に知覚される。

先ず、外部マイク１６ａ，１６ｂが、主として目標からの外部の音を集音してそれぞれアナログ音響信号を出力する。ここで、外部マイク１６ａ，１６ｂはロボット内部からのノイズも集音するが、外装１４自体が頭部１３の内部を密閉しており、また外部マイク１６ａ，１６ｂが頭部１３の内部に対して遮音されていることにより、アナログ音響信号に混入するノイズは、比較的低いレベルに抑えられている。

これに対して、内部マイク１７ａ，１７ｂは、主としてロボット内部からのノイズ、例えば前述した各駆動機構の作動音や冷却ファンの作動音等のノイズ発生源からのノイズを集音する。ここで、内部マイク１７ａ，１７ｂは、外部からの音も集音するが、外装１４が内部を密閉していることから、混入するノイズのレベルは比較的低く抑えられている。

このようにして集音された外部マイク１６ａ，１６ｂからのアナログ音響信号

及び内部マイク１７ａ，１７ｂからのアナログ音響信号は、それぞれアンプ２１ａ乃至２１ｄにより増幅された後、ＡＤコンバータ２２ａ乃至２２ｄによりデジタル音響信号ＳＯＬ，ＳＯＲ，ＳＩＬ，ＳＩＲにＡＤ変換されて、ノイズ除去回路２３及び２４に入力される。

ノイズ除去回路２３，２４は、それぞれ外部マイク１６ａ，１６ｂからの音響信号ＳＯＬ，ＳＯＲから内部マイク１７ａ，１７ｂからの音響信号ＳＩＬ，ＳＩＲを減算する等の演算処理を行なうことにより、外部マイク１６ａ，１６ｂからの音響信号ＳＯＬ，ＳＯＲから、ロボット内部のノイズ発生源からのノイズ信号を除去する。これにより、ノイズ除去回路２３，２４はそれぞれノイズを除去した真の音響信号ＳＬ，ＳＲを外部に出力する。

そして、ピッチ抽出部２５，２６が、それぞれ音響信号ＳＬ，ＳＲに基づいて周波数解析により音響データＤＬ，ＤＲを生成する。

ここで、音響データは、例えば図９に示すようなスペクトログラムとして示される。図９の音響データにおいて、横軸を時刻ｔ（秒）、縦軸を周波数（Ｈｚ）とし、濃度をパワーとして、例えば時刻ｔ１から時刻ｔ２まで断続的に連なった周波数成分が示されている。この周波数成分は、図示の場合、１０００Ｈｚ，２０００Ｈｚ，３０００Ｈｚ，・・・のピークの連なりを有しており、これらのピークがそれぞれ１０００Ｈｚの整数倍であることから、一般的には、これらの周波数成分が唯一つの音を表わし、あるいはそれぞれ一つの音を表わすものと解され得る。この場合には、これらの周波数成分が、同時刻に開始し且つ同時刻に終了していることから、これらの周波数成分が唯一つの音の調波構造を示していると判断される。従って、ピッチ抽出部２５，２６は、最も低い周波数である基本周波数１０００Ｈｚをピッチとする。

このようにして、ピッチ抽出部２５，２６は、音響信号ＳＬ，ＳＲに含まれる全ての音に関して当該音に関するピッチを抽出し、このピッチに対応する当該音の調波構造，開始時ｔ１，終了時ｔ２と共に、音響データＤＬ，ＤＲ，ピッチ情報として左右チャンネル対応部２７に出力する。

続いて、左右チャンネル対応部２７が、これらの音響データＤＬ，ＤＲに基づいて音源毎の音の方向付けを行なう。

この場合、左右チャンネル対応部２７は、例えば各ピッチ抽出部２５，２６で抽出された音響データＤＬ，ＤＲに基づいて、左右のチャンネルの調波構造を比較して最も近いピッチを対応付ける。その際、左右のチャンネルのピッチを一対一で比較するだけでなく、一方のチャンネルの複数のピッチを他方のチャンネルの一つのピッチと比較する等により、より正確な対応付けを行なうことが好ましい。そして、左右チャンネル対応部２７は、対応付けられたピッチの位相を比較することにより音源の方向を容易に演算して方向付けることができる。

これにより、音源分離部２８が、左右チャンネル対応部２７からの音の方向付け情報に基づいて、音響データＤＬ，ＤＲから音源毎の音に関する音響データを抽出して、音源毎の音に分離する。このようにして、聴覚システム２０は音源毎の音の分離によって音響認識を行ない、能動聴覚を行なうことができる。

ここで、内部マイク１７ａ，１７ｂは、それぞれ取付金具４０，４１に固定保持されると共に、これらの取付金具４０，４１が支持部材４２，４３で固定保持されることにより、カメラ１５に対して固定保持されている。

さらに、取付金具４０，４１は、それぞれ支持部材４２，４３のスロット４２ａ，４３ａに挿通された固定ネジ４５，４６により支持部材４２，４３に対して螺着されている。

その際、取付金具４０，４１そして内部マイク１７ａ，１７ｂは、固定ネジ４５，４６が緩められた状態にて、固定ネジ４５，４６がそれぞれ支持部材４２，４３のスロット４２ａ，４３ａ内を摺動することにより、横方向の位置が適宜に調整され、調整後に固定ネジ４５，４６が締め付けられる。

これにより、内部マイク１７ａ，１７ｂは、頭部１３の外装１４の内側にて横方向に移動調整されることにより、所望の位置に設定され得る。即ち、内部マイク１７ａ，１７ｂは、それぞれ対応する外部マイク１６ａ，１６ｂに対して適宜の距離の位置に調整されることによって、前述したノイズ除去回路２３，２４によるノイズキャンセルを最も効果的に行なうことができる。

このようにして、本発明実施形態による人型ロボット１０によれば、内部マイク１７ａ，１７ｂを横方向に適宜に調整した状態にて、ノイズ除去回路２３，２４により、外部マイク１６ａ，１６ｂからの音響信号ＳＯＬ，ＳＯＲから、内部

マイク１７ａ，１７ｂからの音響信号ＳＩＬ，ＳＩＲに基づいてノイズキャンセルを行なうことにより、各駆動機構によって外部マイク１６ａ，１６ｂの指向性の方向を目標に向けながら、Ｓ／Ｎ比の良好な音響信号を得て、各音源からの音を分離してより正確な音声認識を行なうことができる。

従って、例えば目標が移動している場合であっても、各駆動機構によって外部マイク１６ａ，１６ｂの指向性の方向を常に目標に対して追従させながら、目標の音響認識を行なうことができる。

このとき、左右チャンネル対応部２７が、例えば他の知覚システム３０として視覚システムからの視覚による目標の方向付け情報を参照して、音の方向付けを行なうことにより、より正確な音の方向付けを行なうことができる。

また、他の知覚システム３０として視覚システムが使用される場合、左右チャンネル対応部２７が視覚システムに対して音の方向付け情報を出力するようにしてもよい。これにより、視覚システム３０が、画像認識により目標の方向付けを行なう際に、聴覚システム２０からの音に関する方向付け情報を参照することによって、例えば目標が移動して障害物の裏に隠れてしまった場合でも、目標からの音を参照して、目標の方向付けをより一層正確に行なうことができる。

図１０は上述した内部マイク１７ａ，１７ｂの取付構造の変形例を示している。図１０において、内部マイク１７ａ，１７ｂを横方向に移動可能に支持する支持部材４２，４３が、さらにカメラ１５の上面に対して図示しない適宜の手段で固定された第二の支持部材４７，４８に対して、前後方向に摺動可能に取り付けられている。上記第二の支持部材４７，４８はそれぞれ前後方向に延びるスロット４７ａ，４８ａを備えている。

そして、上記支持部材４２，４３の両端に形成された垂直に延びる折曲部４２ｂ，４２ｃ（４３ｂ，４３ｃ）が、それぞれこれらのスロット４７ａ，４８ａに挿通された固定ネジ４９，５０を受容するネジ孔（図示せず）を有しており、これらの固定ネジ４９，５０が締め付けられることにより、第二の支持部材４７，４８に対して固定保持される。

このような構成の内部マイク１７ａ，１７ｂの取付構造によれば、内部マイク１７ａ，１７ｂはそれぞれ取付金具４０，４１に固定保持されると共に、これら

の取付金具４０，４１が支持部材４２，４３及び第二の支持部材４７，４８で固定保持されることにより、カメラ１５に対して固定保持されている。

さらに、取付金具４０，４１は、それぞれ支持部材４２，４３のスロット４２ａ，４３ａに挿通された固定ネジ４５，４６で支持部材４２，４３に対して螺着されており、また支持部材４２，４３は、それぞれ第二の支持部材４７，４８のスロット４７ａ，４８ａに挿通された固定ネジ４９，５０で第二の支持部材４７，４８に対して螺着されている。

その際、取付金具４０，４１そして内部マイク１７ａ，１７ｂは、固定ネジ４５，４６が緩められた状態にて、固定ネジ４５，４６がそれぞれ支持部材４２，４３のスロット４２ａ，４３ａ内を摺動することにより横方向の位置が適宜に調整され、調整後に固定ネジ４５，４６が締め付けられる。

また、上記支持部材４２，４３は、固定ネジ４９，５０が緩められた状態にて、固定ネジ４９，５０がそれぞれ第二の支持部材４７，４８のスロット４７ａ，４８ａ内を摺動することにより前後方向の位置が適宜に調整され、調整後に固定ネジ４９，５０が締め付けられる。

これにより、内部マイク１７ａ，１７ｂは、頭部１３の外装１４の内側にて横方向及び前後方向に移動調整されることにより、所望の位置に設定され得る。即ち、内部マイク１７ａ，１７ｂは、それぞれ対応する外部マイク１６ａ，１６ｂに対して適宜の距離の位置に調整されることによって、前述したノイズ除去回路２３，２４によるノイズキャンセルをより一層効果的に行なうことができる。

上述した実施形態において、人型ロボット１０は、４ＤＯＦ（自由度）を有するように構成されているが、これに限らず、任意の動作を行なうように構成されたロボットに本発明によるロボット聴覚システムを組み込むことも可能である。また、上述した実施形態においては、本発明によるロボット聴覚システムを人型ロボット１０に組み込んだ場合について説明したが、これに限らず、犬型等の各種動物型ロボットや、その他の形式のロボットに組み込むことも可能であることは明らかである。

さらに、上述した実施形態において、内部マイク１７は一対のマイク１７ａ，１７ｂから構成されているが、二対以上のマイクから構成されていてもよい。

さらにまた、上述した実施形態においては、外部マイク１６は、一対のマイク１６ａ，１６ｂから構成されているが、二対以上のマイクから構成されていてもよい。

また、上述した実施形態では、内部マイク１７ａ，１７ｂは、支持部材４２，４３のスロット４２ａ，４３ａに沿って横方向に移動調整可能に構成されているが、これに限らず他の手段によって横方向に移動調整可能に構成されていてもよい。

取付金具４０，４１は、駆動手段によって支持部材４２，４３そしてカメラ１５に対して横方向に移動調整可能に構成されていてもよい。この場合、人型ロボット１０の頭部１３から外装１４を外すことなく、組み立てた状態にて内部マイク１７ａ，１７ｂの横方向の位置調整を行なうことができるので、ノイズキャンセルをより効果的に、そしてより正確に行なうことができる。

さらに、上述の実施形態においては、内部マイク１７ａ，１７ｂは、横方向にあるいは横方向及び前後方向に移動可能に支持されているが、これに限らず、さらに上下方向に移動可能に支持されていてもよい。この場合、内部マイク１７ａ，１７ｂは、公知の構成の三軸方向に移動調整可能な機構により支持される。

以上述べたように、本発明によれば、外部マイクが主として外部の目標からの音を集音し、内部マイクが主としてロボット内部の駆動機構等のノイズ発生源からのノイズを集音する。その際、外部マイクが集音した音響信号には、ロボット内部のノイズ発生源からのノイズ信号が混入しているが、この混入したノイズ信号は、ノイズ除去部における演算処理によって内部マイクが集音したノイズ信号にてキャンセルされることにより著しく低減される。

そして、ノイズがキャンセルされた音響信号から、ピッチ抽出部により周波数解析によって音響信号を取り出して、この音響信号から左右チャンネル対応部により音の方向付けを行なった後、この音の方向付けに基づいて、音源分離部によって各音源毎の音響データに分離する。

したがって、外部マイクからの音響信号は、ノイズ除去部での演算処理によってロボット内部の駆動機構等のノイズ発生源からのノイズが容易に著しく低減されて、そのＳ／Ｎ比が大幅に改善されるので、音源毎の音響データの分離をより

一層良好に行なうことができる。

このようにして、本発明によれば、駆動機構等のロボット内部で発生するノイズに影響されることなく、外部の目標からの音を集音して能動知覚を行なうことができるようにした、極めて優れたロボット聴覚システムが提供される。

さらに、上記内部マイクが外装の内側にて横方向に移動可能に配設されていることにより、ロボット内部のノイズ発生源からのノイズをより良好に集音するように内部マイクを横方向に移動させて、各内部マイクと対となるべき対応する外部マイクとの距離を適宜に調整することができ、外部マイクが集音する音響信号からより良好にノイズをキャンセルすることができる。

なお、本発明は例示的な実施例について説明したものであり、本発明の要旨及び範囲を逸脱することなく、実施例での種々の変更、省略、追加が可能であることは当業者において自明である。したがって、本発明は実施例に限定されるものではなく、請求の範囲に記載された要素に基づいて規定される範囲及び及びその均等範囲を包含するものとして理解されなければならない。

産業上の利用可能性

以上のように、本発明によるロボット聴覚システムは、ロボット、特に人型または動物型ロボットにおける能動聴覚のための聴覚システムとして極めて有用である。

請 求 の 範 囲

1．内部にノイズ発生源を備えたロボットにおいて、

少なくともロボットの一部を覆う防音性の外装と、

上記外装の外側に設けられ、主として外部の音を集音する少なくとも一対の外部マイクと、

上記外装の内側に設けられ、主として内部のノイズ発生源からのノイズを集音する少なくとも一対の内部マイクと、

上記外部マイク及び内部マイクからの左右の信号に基づいて、それぞれ外部マイクからの音響信号から内部のノイズ発生源からのノイズ信号をキャンセルするノイズ除去部と、

上記ノイズ除去部からの左右の音響信号からそれぞれ周波数解析を行って、時間，周波数及びパワーに関する音響データ及びピッチ情報を取り出すピッチ抽出部と、

上記ピッチ抽出部で取り出された左右の音響データ及びピッチ情報に基づいて、当該音響データが示すピッチを伴う調波構造から音の方向付けを行なう左右チャンネル対応部と、

上記左右チャンネル対応部で取り出された音の方向付け情報に基づいて、上記音響データから音源毎の音響データに分離する音源分離部と、を含んでいることを特徴とする、ロボット聴覚システム。

2．内部に駆動機構等のノイズ発生源を備えた人型または動物型のロボットにおいて、

少なくともロボットの頭部を覆う防音性の外装と、

上記外装の外側にて両側の耳に対応する両耳位置に設けられ、主として外部の音を集音する少なくとも一対の外部マイクと、

上記外装の内側に設けられた、主として内部のノイズ発生源からのノイズを集音する少なくとも一対の内部マイクと、

上記外部マイク及び内部マイクからの信号に基づいて外部マイクからの音響

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信号から内部のノイズ発生源からのノイズ信号をキャンセルするノイズ除去部と、

上記ノイズ除去部からの左右の音響信号からそれぞれ周波数解析を行って、時間，周波数及びパワーに関する音響データ及びピッチ情報を取り出すピッチ抽出部と、

上記ピッチ抽出部で取り出された左右の音響データ及びピッチ情報に基づいて、当該音響データのピッチを伴う調波構造から音の方向付けを行なう左右チャンネル対応部と、

上記左右チャンネル対応部で取り出された音の方向付け情報に基づいて、上記音響データから音源毎の音響データに分離する音源分離部と、を含んでいることを特徴とする、ロボット聴覚システム。

3．さらに、ロボットが視覚，触覚等の知覚システムを備えており、前記左右チャンネル対応部が、これらの知覚システムからの情報を参照して音の方向付けを行なうことを特徴とする、請求の範囲第1項又は第2項に記載のロボット聴覚システム。

4．前記左右チャンネル対応部が、前記知覚システムに対して音の方向付けに関する情報を出力することを特徴とする、請求の範囲第3項に記載のロボット聴覚システム。

5．前記内部マイクが、外装の内側にて少なくとも横方向に移動可能に配設されていることを特徴とする、請求の範囲第1項から第4項の何れかに記載のロボット聴覚システム。

6．前記内部マイクが、外装の内側にて横方向に延びる支持部材に対して摺動可能に支持されていることを特徴とする、請求の範囲第5項に記載のロボット聴覚装置。

7．前記支持部材が横方向に延びるスロットを備えており、前記内部マイクが、

上記支持部材のスロットに沿って摺動可能に支持された可動部材に固定されていることを特徴とする、請求の範囲第６項に記載のロボット聴覚装置。

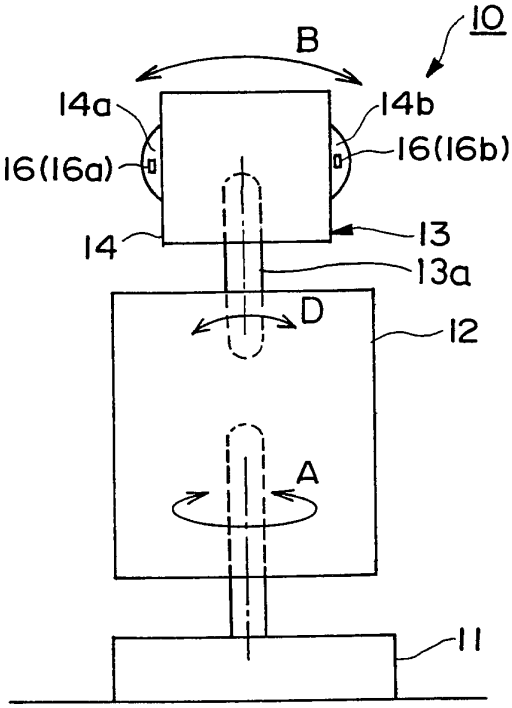
８．前記可動部材がスロットに挿通された固定ネジにより支持部材に対して螺着されることを特徴とする、請求の範囲第７項に記載のロボット聴覚装置。

９．前記内部マイクが、さらに前後方向に移動可能に配設されていることを特徴とする、請求の範囲第５項に記載のロボット聴覚装置。

１０．前記内部マイクが、外装の内側にて横方向に延びる第一の支持部材に対して摺動可能に支持されていて、この第一の支持部材が、外装の内側にて前後方向に延びる第二の支持部材に対して摺動可能に支持されていることを特徴とする、請求の範囲第９項に記載のロボット聴覚装置。

１１．前記内部マイクが、さらに上下方向に移動可能に配設されていることを特徴とする、請求の範囲第５項、第９項及び第１０項の何れかに記載のロボット聴覚装置。

第 1 図



第 2 図

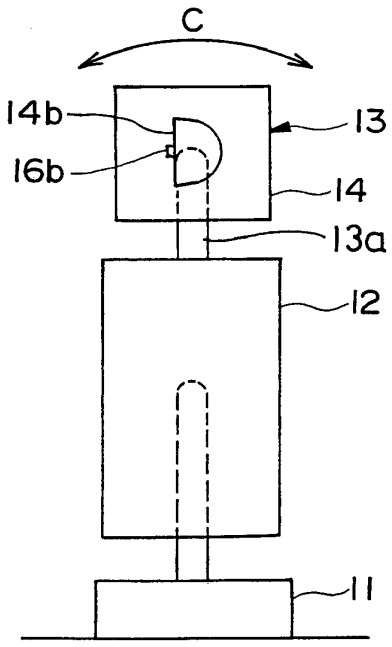
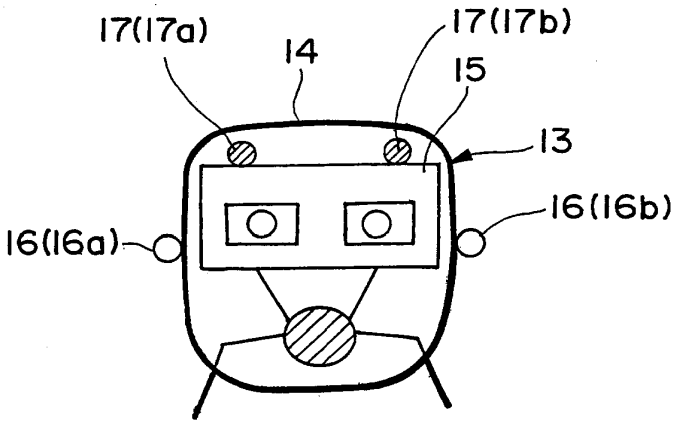


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第 3 図



第 4 図

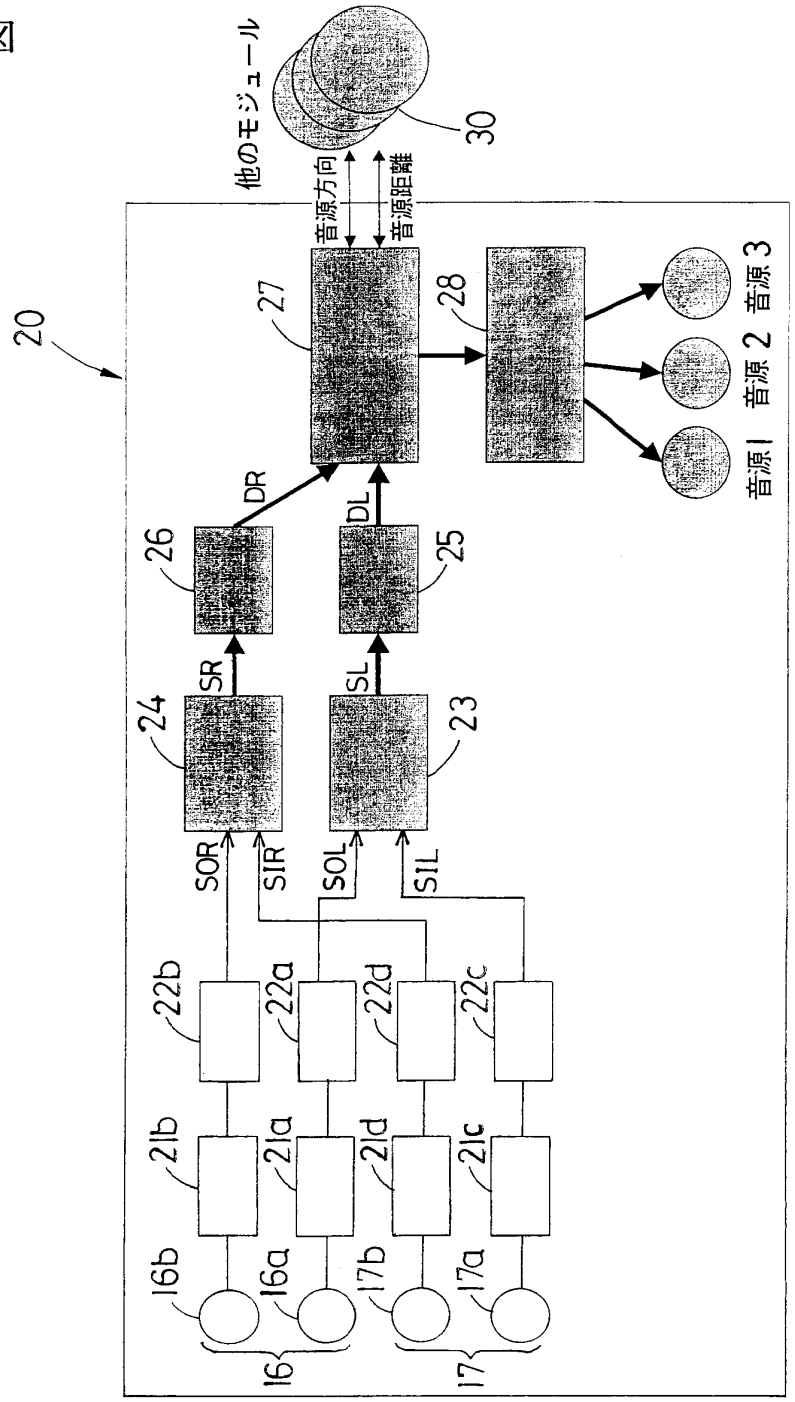
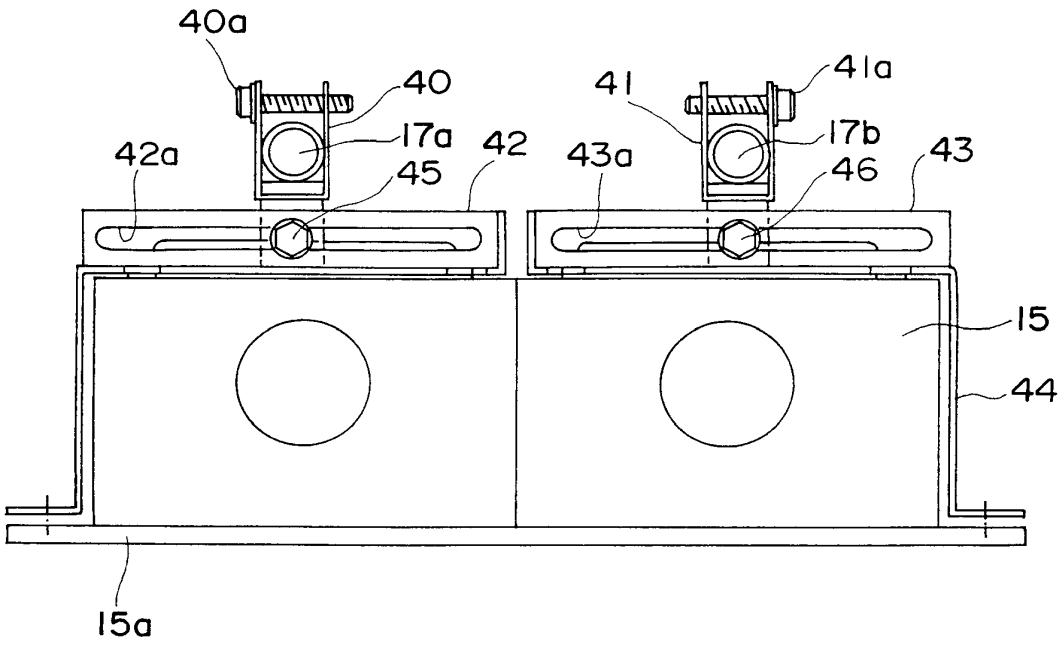
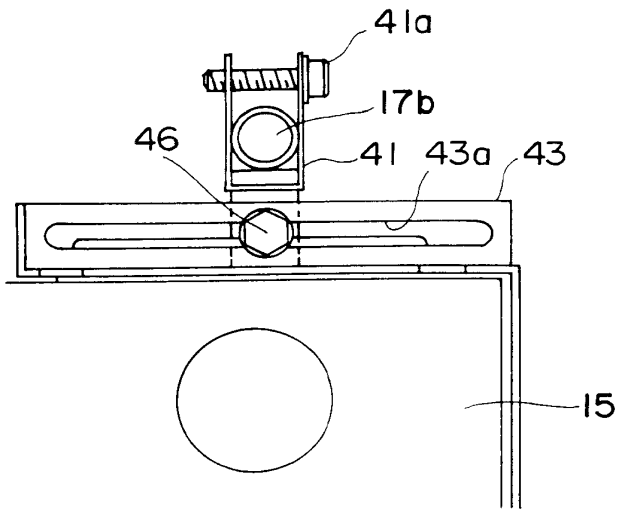


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第 5 図



第 6 図



第 7 図

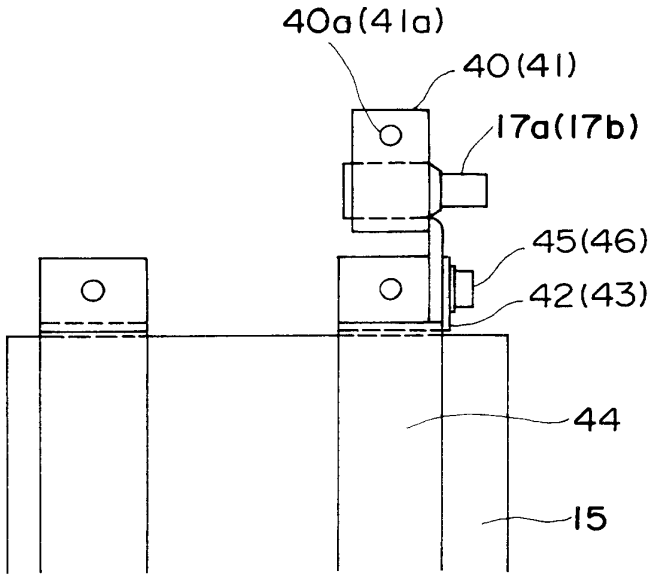
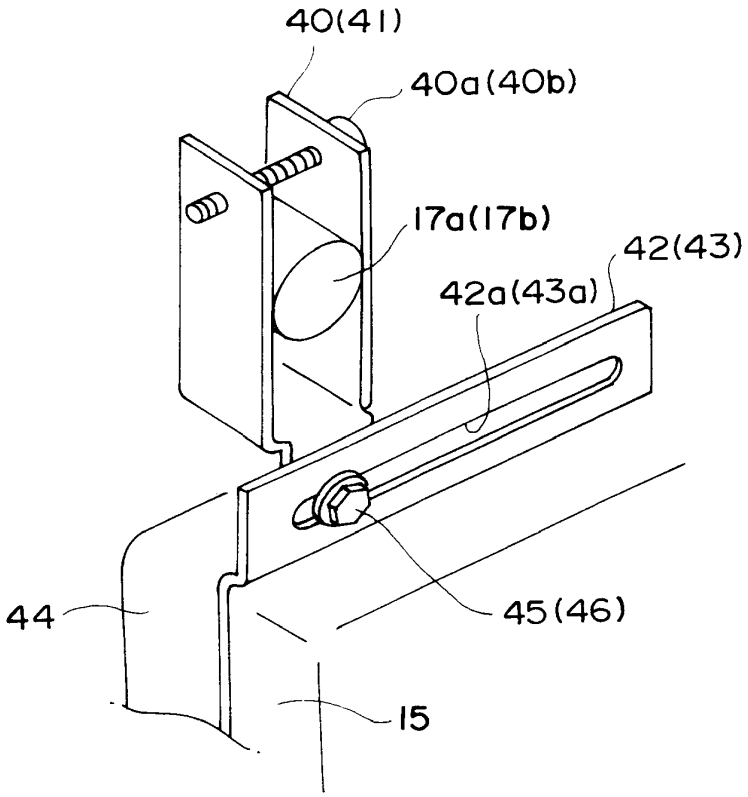
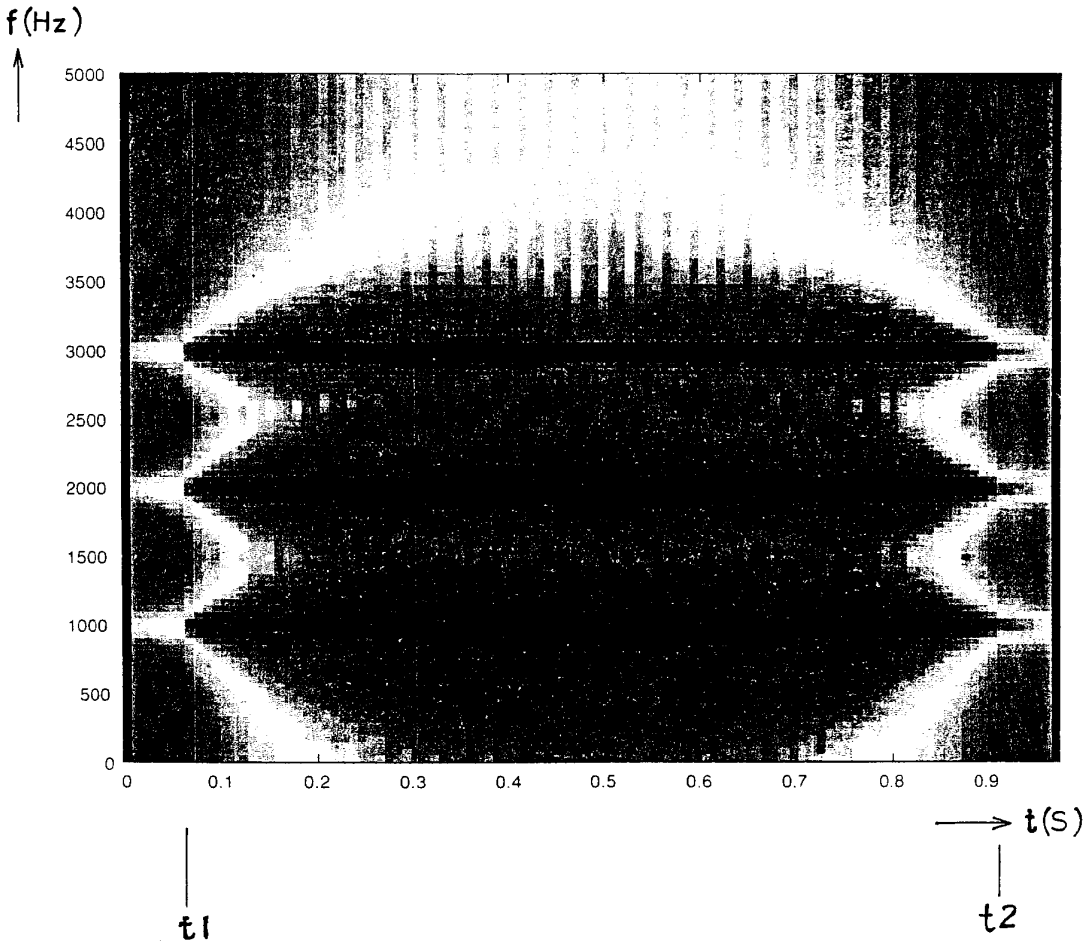


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Page 1828 of 2198

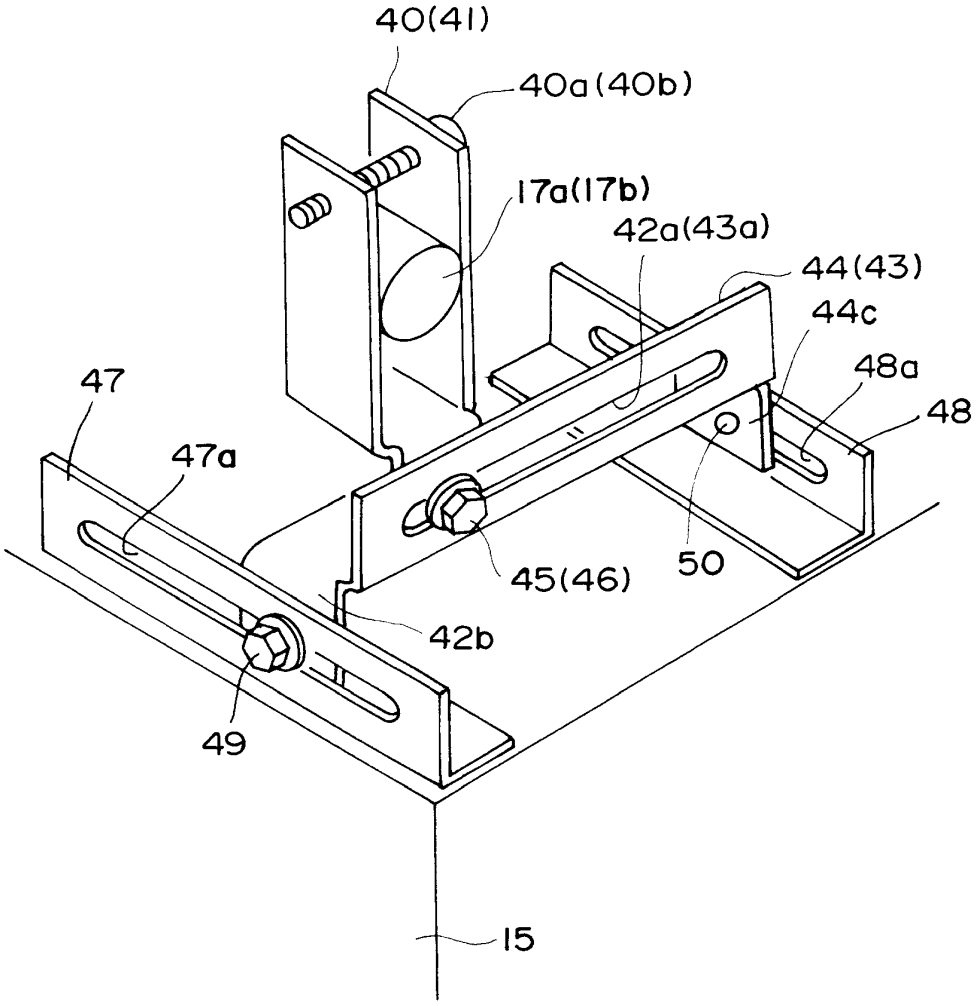
第 8 図



第 9 図


f(Hz)
5000
4500
4000
3500
3000
2500
2000
1500
1000
500
0
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
t(S)
t1
t2

第 10 図


40(41)
40a(40b)
17a(17b)
42a(43a)
44(43)
44c
48a
48
47
47a
45(46)
50
42b
49
15

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP01/00647

A. CLASSIFICATION OF SUBJECT MATTER
Int.Cl[7] G10L21/02, 15/20, B25J13/00

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)
Int.Cl[7] G10L21/02, 15/20

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
Jitsuyo Shinan Koho 1922-1996 Toroku Jitsuyo Shinan Koho 1994-2001
Kokai Jitsuyo Shinan Koho 1971-2001 Jitsuyo Shinan Toroku Koho 1996-2001

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	H. G. Okuno et al., "Research Issues of Humanoid Audition", JSAI Technical Report, Proceedings of the Seventh Meeting of Special Interest Group on AI Challenges, SIG-Challenge-9907-10, Japanese Society for Artificial Intelligence, pp.61-65, 02 November,1999 (02.11.99)	1-11
Y	T. Kikuchi et al., "Microphone Array System with DOA Estimation by Using Harmonic Structure of Speech Signals", IEICE Technical Report [Digital Shingo Shori], Vol.98, No.534, DSP98-164, the Institute of Electronics, Information and Communications Engineers, pp.23-28, 22 January, 1999 (22.01.99),	1-11
A	S. Nakamura et al., "Speech Recognition with Source Localization by Microphone Array", the Heisei 7 Spring Meeting of the Acoustical Society of Japan, the Acoustical Society of Japan, Vol.1, 1-5-8, pp.15-16, 14 March, 1995 (14.03.95),	1-11

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

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search
21 March, 2001 (21.03.01)

Date of mailing of the international search report
03 April, 2001 (03.04.01)

Name and mailing address of the ISA/
Japanese Patent Office

Facsimile No.

Authorized officer

Telephone No.

Form PCT/ISA/210 (second sheet) (July 1992)

国際調査報告

国際出願番号　ＰＣＴ／ＪＰ０１／００６４７

A.　発明の属する分野の分類（国際特許分類（ＩＰＣ））

Int Cl' Ｇ１０Ｌ２１／０２，１５／２０，Ｂ２５Ｊ１３／００

B.　調査を行った分野

調査を行った最小限資料（国際特許分類（ＩＰＣ））

Int Cl' Ｇ１０Ｌ２１／０２，１５／２０

最小限資料以外の資料で調査を行った分野に含まれるもの
日本国実用新案公報　１９２２～１９９６年
日本国公開実用新案公報　１９７１～２００１年
日本国登録実用新案公報　１９９４～２００１年
日本国実用新案登録公報　１９９６～２００１年

国際調査で使用した電子データベース（データベースの名称、調査に使用した用語）

ＪＩＣＳＴ科学技術文献ファイル（ＪＯＩＳ）

C.　関連すると認められる文献

引用文献のカテゴリー＊	引用文献名　及び一部の箇所が関連するときは、その関連する箇所の表示	関連する請求の範囲の番号
Y	人工知能学会研究会資料(JSAI Technical Report), AIチャレンジ研究会(第７回)(Proceedings of the Seventh Meeting of Special Interest Group on AI Challenges), SIG-Challenge-9907-10, H.G.Okuno et al., "Research Issues of Humanoid Audition", p.61-65, 2.11月.1999(02.11.99), 社団法人人工知能学会(Japanese Society for Artificial Intelligence)	1-11

☒ Ｃ欄の続きにも文献が列挙されている。　☐ パテントファミリーに関する別紙を参照。

＊　引用文献のカテゴリー
「Ａ」特に関連のある文献ではなく、一般的技術水準を示すもの
「Ｅ」国際出願日前の出願または特許であるが、国際出願日以後に公表されたもの
「Ｌ」優先権主張に疑義を提起する文献又は他の文献の発行日若しくは他の特別な理由を確立するために引用する文献（理由を付す）
「Ｏ」口頭による開示、使用、展示等に言及する文献
「Ｐ」国際出願日前で、かつ優先権の主張の基礎となる出願の日の後に公表された文献
「Ｔ」国際出願日又は優先日後に公表された文献であって出願と矛盾するものではなく、発明の原理又は理論の理解のために引用するもの
「Ｘ」特に関連のある文献であって、当該文献のみで発明の新規性又は進歩性がないと考えられるもの
「Ｙ」特に関連のある文献であって、当該文献と他の１以上の文献との、当業者にとって自明である組合せによって進歩性がないと考えられるもの
「＆」同一パテントファミリー文献

国際調査を完了した日	国際調査報告の発送日
２１．０３．０１	03.04.01

国際調査機関の名称及びあて先	特許庁審査官（権限のある職員）	５Ｃ	８８４２
日本国特許庁（ＩＳＡ／ＪＰ） 郵便番号１００－８９１５ 東京都千代田区霞が関三丁目４番３号	松尾　淳一　印		
	電話番号　０３－３５８１－１１０１　内線　３５４０		

様式ＰＣＴ／ＩＳＡ／２１０（第２ページ）（１９９８年７月）

Ｃ（続き）．　関連すると認められる文献

引用文献のカテゴリー＊	引用文献名　及び一部の箇所が関連するときは、その関連する箇所の表示	関連する請求の範囲の番号
Y	電子情報通信学会技術研究報告("IEICE Technical Report")[ディジタル信号処理], Vol.98, No.534, DSP98-164, 菊池隆文外("T.Kikuchi et al."), 「音声の調波構造を方向推定に適用したマイクロフォンアレイシステム」("Microphone Array System with DOA Estimation by Using Harmonic Structure of Speech Signals"), p.23-28, 22.1月.1999(22.01.99), 社団法人電子情報通信学会(The Institute of Electronics, Information and Communications Engineers)	1-11
A	日本音響学会平成７年度春季研究発表会講演論文集Ｉ("The Heisei 7 Spring Meeting of the Acoustical Society of Japan, Vol.1"), 1-5-8, 中村哲外("S.Nakamura et al."), 「マイクロホンアレーを用いた音源方向検出による音声認識」("Speech Recognition with Source Localization by Microphone Array"), p.15-16, 14.3月.1995(14.03.95), 社団法人日本音響学会("The Acoustical Society of Japan")	1-11

(11) Application No. AU 2006200446 A1

(19) AUSTRALIAN PATENT OFFICE

(54) Title
Ear Microphone Apparatus and Method

(51)[6] International Patent Classification(s)
A61B 5/16 (2006.01) 20060101AFI20060207
A61B 5/16 BHAU

(21) Application No: **2006200446** (22) Application Date: **2006 .02 .02**

(43) Publication Date : **2006 .02 .23**
(43) Publication Journal Date : **2006 .02 .23**

(62) Divisional of:
2002307038

(71) Applicant(s)
Think-A-Move, Ltd

(72) Inventor(s)
Nemirovski, Guerman G.

(74) Agent/Attorney
Cullen & Co, Level 26 239 George Street, Brisbane, QLD, 4000

2006200446 02 Feb 2006

ABSTRACT

An earset including a housing positionable with respect to an ear of a person, a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure and a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal. The earset also includes a circuit coupled to receive the microphone signal and the speaker signal and operable to produce a corrected microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker to produce a corrected microphone signal.

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AUSTRALIA
Patents Act 1990

COMPLETE SPECIFICATION FOR A STANDARD PATENT

Names of Applicant:	Think-A-Move, Ltd.
Address for Service:	CULLEN & CO. Level 26 239 George Street Brisbane Qld 4000
Invention Title:	Ear Microphone Apparatus and Method

The following statement is a full description of the invention, including the best method of performing it, known to us:

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FIELD OF THE INVENTION

The present invention generally relates to a sensor placed with respect to an ear. More particularly, the sensor is used to detect physical or mental actions of a person such as speech, thoughts, movements of the tongue with respect to the oral cavity, biological functions, etc. and provides an output which can be converted into system control functions. Furthermore, the sensor is used to detect speech and other oral vocalizations of a person with a microphone as well as to deliver sound to the ear of the person with a speaker, while feedback from the speaker to the microphone is reduced.

BACKGROUND OF THE INVENTION

An apparatus and method for detecting thoughts is disclosed in U.S. Patent No. 6,024,700, the entire disclosure of which is incorporated by reference. The thought may lead to a detectable air pressure or sound at one or both ears of a user which can be detected by a detector, such as a microphone. Exemplary outputs from such system and method for detecting thoughts may be one or plural discrete outputs, such as discrete outputs representing respective alphanumeric characters, directions, etc.

An apparatus and method for detecting physical action, such as mouth and tongue movement, is disclosed in U.S. Patent Application Serial No. 09/706,091, filed November 3, 2000, the entire disclosure of which is incorporated by reference. The physical action leads to a detectable air pressure or sound at one or both ears of a user.

Sounds produced by a person's ears sometimes are referred to as autoaccoustic. Autoaccoustic sounds have been used in the past for diagnostics, e.g., to indicate ear function or a condition of the ear. For example, in response to a sound input to the ear, the ear has been found to provide a detectable autoaccoustic output.

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At present there are available controllers to enable a handicapped person, who cannot use arms or legs, to operate a motorized wheelchair. Such a controller includes a track-ball or joystick type device that is inserted into the mouth of an individual; and the individual operates the controller using mouth and/or tongue movements to provide mechanical inputs. Such device is difficult to operate and is problematic if it falls from the mouth or irritates the mouth.

Other controllers exist to enable handicapped individuals to perform functions with mechanical assists. Operation of such controllers typically requires a signal, such as an incident electromagnetic signal, e.g., infrared or ultraviolet light/radiation, which is directed at the individual; then, by a specified action or activity the individual can modulate the incident electromagnetic signal, e.g., by blinking the eyes, moving the head, or some such action. The need for incident electromagnetic signal is disadvantageous for several reasons: an extra piece of equipment is required to provide that signal, the signal must be properly directed or aligned, the detector and the modulated signal must be properly aligned, the signal should not have a detrimental affect on the user (such as causing eye injury), etc.

At present there are available voice recognition software programs which receive an input signal representative of human speech and convert that signal into text format or carry out a command upon interpreting the speech. The input signal is generated by a microphone disposed in front of the speaker's mouth.

Furthermore, wireless mobile telephones, also referred to as a cellular telephones, have become exceedingly popular communication devices. However, the vast majority of cellular telephones contain a transceiver (e.g., a radio frequency , or RF, transceiver) for establishing a communication link with a remote location, such as a cell phone tower. In order to carry out a conversation with another person using the wireless telephone, the user must hold the telephone adjacent the ear and mouth of the user. This presents the disadvantage of occupying the use of at least one of the user's hands. In many situations, hands free use of the wireless telephone is desirable. The same is true for receivers, or handsets, found in hard-wired telephone systems. For example, the user of a hard-wired telephone system may wish to type on a computer while speaking on the phone. In addition, medical professionals and others have expressed concerns relating to the health of wireless telephone users who engage in prolonged use of an RF transceiver adjacent their head.

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There are many commercially available headsets available to users of wireless and/or hard-wired telephone systems. These headsets are intended to assist the user in carrying out a conversation without the use of the user's hands and to locate the telephone (and any RF transceiver) away from the user's head. These headsets typically include an ear piece containing a speaker. The ear piece can be removably placed with respect to the user's ear and broadcasts sounds to the user's ear. The headsets also typically include a microphone disposed on a support member that positions the microphone with respect to the user's mouth. The microphone is used to detect speech and other vocalizations emanating from the mouth of the user. The detected sounds are converted into an electrical signal and transmitted by the telephone to a backbone telecommunications network and on to the telephone of another person. In this manner the user can carry out a fully duplexed conversation with the other person.

However, the headsets can be cumbersome to use. More particularly, care must be taken to ensure that the microphone is properly positioned and that the microphone maintains that position. The need to adjust the headset during a conversation can be distracting to the user. In addition, improper positioning of the microphone may lead to poor and/or unreliable detection of the user's speech. This problem is compounded by the common occurrence of the microphone detecting environmental noise, such as the sound of a passing vehicle, conversations taking place near the user and the like. The detected environmental noise is ultimately transmitted by the telephone.

Accordingly, there exists a need in the art for an easy to use telephone headset that accurately detects the user's speech, even when speaking at a low volume, and reduces the detection of environmental noise.

SUMMARY OF THE INVENTION

According to a first aspect of the invention, there is provided an earset comprising:

a housing positionable with respect to an ear of a person;

a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure;

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a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal input to the speaker; and

a circuit coupled to receive the microphone signal produced by the microphone and the electrical speaker signal as input to the speaker, and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker wherein the circuit includes:

an amplifier for amplifying the microphone signal produced by the microphone;

an amplifier for amplifying the electrical speaker signal received by the circuit and outputting an electrical feedback indication signal; and

a combining means for combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

According to a second aspect of the invention, there is provided in combination, a communications system including the earset according to the first aspect, and further comprising a communication device coupled to receive the corrected microphone signal and transmit an output corresponding to the corrected microphone signal to a remote location, and the communication device adapted to receive a signal from which the speaker signal is produced by the communication device.

According to a third aspect of the invention, there is provided a method of detecting speech of a person, comprising the steps of:

placing a microphone at least partially within an ear of the person;

detecting a change in air pressure within the ear of the person using the microphone, the change in air pressure corresponding to the speech of the person;

producing an electrical microphone signal corresponding to the internally detected change in air pressure;

placing a speaker with respect to the ear of the person;

generating a sound with the speaker corresponding to an electrical speaker signal input to the speaker; and

generating a corrected microphone signal by reducing a feedback component of the microphone signal, the feedback component resulting from the detection by the

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microphone of the sound generated by the speaker and wherein reducing the feedback includes:

amplifying the microphone signal produced by the microphone;

amplifying the electrical speaker signal to generate an electrical feedback indication signal; and

combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1A is a block level diagram illustrating a system by which air pressure fluctuations are detected and used to provide a control function which corresponds to the detected air pressure according to the present invention;

FIG. 1B is a block level diagram illustrating a system by which air pressure fluctuations are detected and processed, and in the illustrated embodiment, processed by a voice recognition software program to interpret speech of a person;

FIG. 1C is a block level diagram illustrating a system by which air pressure fluctuations are detected and re-transmitted by a communications device;

FIG. 1D is a block level diagram illustrating a system by which air pressure fluctuations are detected and used by a medical diagnostics device;

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FIG. 1E is a block diagram illustrating a system having a sensor with an internal microphone and an external microphone;

FIG. 2A is a schematic view of an ear;

FIG. 2B is a cross-sectional view of a sensor embodiment for monitoring air pressure changes within the ear of a person;

FIG. 2C is a graph illustrating an exemplary electrical signal produced by a sensor according to the present invention;

FIG. 3A is a flow chart diagram illustrating a method of detecting changes in air pressure and providing a control instruction in response to the detected changes according to the present invention;

FIG. 3B is a flow chart diagram illustrating another method of detecting changes in air pressure and providing a control instruction in response to the detected changes according to the present invention;

FIG. 4 is a flow chart diagram illustrating a method of monitoring according to the present invention;

FIG. 5 is a flow chart diagram illustrating a method of processing an electrical signal corresponding to a mouth/tongue action, a thought, speech and/or another type of initiating action;

FIG. 6 is a graph illustrating the conversion of an analog electrical signal to digital signal data according to the present invention;

FIG. 7 is a flow chart diagram illustrating a method of analyzing the digital signal data according to the present invention;

FIG. 8 is a graph illustrating a method of analyzing the digital signal data with the flow chart of FIG. 7 according to the present invention;

FIG. 9 is a flow chart diagram illustrating a method of processing the digital signal data according to the present invention;

FIG. 10 is a system diagram showing a system including parts and functions of the invention;

FIG. 11 is a graph representing exemplary raw signal information detected by a sensor used in the invention to detect air pressure changes;

FIG. 12 is a graph of frequency filtered signal information of FIG. 11, the frequency filtering to remove noise, for example;

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FIG. 13 is a graph representing power spectrum window slicing of the curve of FIG. 12;

FIG. 14 is a graph of such power spectrum;

FIG. 15 is a block diagram illustrating a communications system according to an embodiment of the present invention;

FIG. 16 is a cross-sectional view of an ear microphone assembly, or earset, disposed with respect to the ear of a person, and the ear also shown in cross-section;

FIG. 17 is a block diagram of the ear microphone assembly;

FIG. 18 is a block diagram of an ear microphone assembly according to another aspect of the invention; and

FIG. 19 is a flow chart of a method of processing signals generated by the ear microphone assembly of FIG. 17.

DETAILED DESCRIPTION OF THE INVENTION

The following is a detailed description of the present invention made in conjunction with the attached figures, wherein like reference numerals will refer to like elements throughout. To illustrate the present invention in a clear and concise manner, the drawings may not necessarily be to scale and ceratin features may be shown in somewhat schematic form.

One embodiment of the present invention is directed to an apparatus and method for detecting physical and/or mental actions of a person (such as actions occurring in association with the thoughts of the person, the speech of the person and/or movements associated with the oral cavity) by monitoring air pressure changes in or near a person's ear over a period of time. As an example, motion of the tongue in the oral cavity can be detected or sensed to provide a useful output. As another example, the action can be mental, as in thinking a thought, which produces a detectable air pressure change within and/or adjacent the ear as discussed in greater detail in U.S. Patent No. 6,024,700. More specifically, the thought causes a reaction in the human anatomy which effectuates a change in the ear (perhaps by muscle contraction), thereby producing a detectable change in air pressure in or near the ear canal. An exemplary output may contain information analogous to an information signal produced by a computer mouse, the click buttons on the mouse, a computer joystick, or the like.

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The present invention has been found to be more aesthetically pleasing and comfortable to the user and detects less environmental, or ambient, noise than the systems disclosed in the above-mentioned U.S. Patent No. 6,024,700 and in U.S. Patent Application Serial No. 09/706,091. As will be described in more detail below, the sensor of the present invention includes a microphone disposed with respect to an ear or within an ear of a user and detects air pressure changes within the ear. This microphone will be referred to herein as an internal microphone. The air pressure changes may be caused by one or more factors, including vibrations of the ear drum, vibrations of bone within the ear, vibrations of other anatomical structures within the ear and vibrations conducted by the bones and/or tissue of the user to the ear and which invoke an air pressure change in the ear. As a result, the sensor can be used to detect a person's speech when the person is speaking at a normal level or at an extremely low volume (e.g., at a decibel, or volume, level lower than that which is ordinarily not sufficiently detectable by a microphone placed in front of the person's mouth or detectable by another individual located near the person speaking). The term speech, as used herein, is intended to include spoken words and utterances, singing and other vocalizations produced by a person. Such detected speech can be used for many purposes including, for example, processing the detected speech with voice recognition software or re-transmitting the detected speech with a communications device such as a cellular telephone (thereby allowing separation of the speech input device from the RF transceiver).

It is noted that the term air pressure changes is used in its broadest sense and includes, for example, sound waves (whether audible to the user or not), pressure fluctuations, vibrations, resonations and the like. In addition, the term air pressure changes as used herein includes vibrations conducted by the bones and tissue of the user that are carried to ear. These conducted vibrations can vibrate a housing that supports the internal microphone leading to sound detection by the internal microphone. The conducted vibrations may also cause anatomical parts of the ear to resonate, and/or amplify sounds, thereby leading to sound detection by the internal microphone. The air pressure changes may be caused by one or more factors, including vibrations of the ear drum, vibrations of bone within the ear, vibrations of other anatomical structures within the ear and vibrations conducted by the bones and/or tissue of the user to the ear and which invoke an air pressure change in the ear.

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In addition to the internal microphone, the present invention can also include an external microphone. The external microphone detects air pressure changes, or sounds, originating outside the user's ear. As described below in more detail, these sounds are converted into electrical signals and compared with electrical signals corresponding to the sounds detected by the internal microphone. The respective signals can be processed to discern air pressure changes within the ear from externally produced ambient noise, speech and the like.

The present invention can also be used as a medical diagnostic tool by detecting and monitoring various biological processes, including a person's heartbeat, breathing and gastronomical events, which cause detectable changes in air pressure in or near the ear. Therefore, by monitoring air pressure adjacent or within the ear, a person's heartbeat, breathing and any irregularities associated therewith can be studied. The sounds detected by the sensor of the present invention can be amplified to a level to be studied by medical personnel and/or processed by a device programmed to analyze such biological functions.

The present invention is able to provide an output having a continuity characteristic, e.g., similar to the signal produced by moving a computer mouse along a surface. The electrical signal produced by the sensor can contain discernable information which was generated by two or three dimensional movements of certain body parts, particularly movements of the tongue, jaw, and oral and nasal cavities. The resulting signal output from the sensor is analogous to the signal produced by a computer joystick that has up, down, left, right, front and back signal outputs. Intensity or other components of the signal can be used as a measure of the action's speed characteristics.

By measuring air pressure in both ears with separate sensors, at least two electrical signals can be produced and processed, resulting in a two dimensional (or stereophonic) information pattern. The information derived from each sensor, along with similarities and differences in the data streams, can be used to discern additional information from the person. In some cases, the additional information will improve the generation of control commands, detection of thoughts, recognition of speech, detection of biological processes and other functions of the systems described herein.

The invention may be used for many additional purposes. One exemplary use is to provide an input to a computer or to some other electrical or electronic device. An exemplary input is an electrical signal, but it will be appreciated that the signal generated by

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the sensor used to detect air pressure changes may be other than electrical, or an electrical signal generated by the sensor can be converted into a form which is not electrical. Example alternative signal types include, for example, an optical signal, a fluidic signal, a radio frequency (RF) signal, etc. The input may control the computer or other device or it may be used for some other purpose. To facilitate and to avoid complexity of the description of the invention herein, the invention is described with respect to a use to provide an electrical input to a computer, for example, a personal computer or some other computer, whether of the digital or analog type, to control the operation of the computer. It will be appreciated that the invention may be used for many other purposes.

One example of a use of the invention in combination with a computer which is controlled using the invention is to provide functions and capabilities for a handicapped person. For example, the invention may be used to enable a handicapped individual or a non-handicapped individual to control operation of a piece of machinery in a manufacturing facility, to control operation of a motorized wheelchair or similar device, to control operation of a video game, etc. These are only examples, and it will be appreciated that the invention may be used for many other purposes.

In the present invention there is no need for the individual to be subjected to a stimulus signal, such as an incident acoustic or electromagnetic signal. Rather, in the present invention, actions of the individual (e.g., a tongue movement involving clicking of the tongue against the teeth or pallet) are used to create air pressure or sounds at one or both ears of the individual, which can be detected or sensed and used as an output of the invention to be provided as an input to a computer or to some other device. Thus, the present invention presents an unstimulated signal, namely a signal that is produced by action of the individual without the need for a stimulus to be directed at the individual.

In accordance with an embodiment of the invention a sensor senses or detects air pressure, a change in air pressure or a sound that is produced by an ear of an individual. These pressure changes may be produced by the person's ear drum or other anatomical structure within the ear. In addition, the pressure changes may originate in other parts of the person's body and conducted to the ear by bone or tissue where a vibration or air pressure change is produced and detected by the sensor.

To facilitate the description herein, the event of sensing or detecting will be referred to as detecting and that which is detected will be referred to as a change within the ear.

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Also, to facilitate the description herein, the action causing the detectable change will be described as an "initiating action." The initiating action can be one or more physical or mental actions, including, but not limited to, thinking a thought, speaking, movement of the tongue in the oral cavity (sometimes referred to as "tongue action"), other movements or actions in or associated with the oral or nasal cavities, clicking or other actions of the tongue against the teeth or pallet, exhaling or inhaling by the lungs, beating of the heart and actions of the nose. As an example, tongue movements lead to the detectable air pressure. These movements can be two dimensional (e.g., up/down, left/right, in/out, etc.) or three dimensional (any combination of front and back, left and right and up and down) which generate a corresponding output by the sensor, even if that output is one dimensional in the case of one sensor or two dimensional in the case of two sensors (one for each ear).

The present invention monitors changes within the human ear which occur instantaneously, or nearly instantaneously, in response to the initiating action to provide a substantially real-time detection and control system. In addition, the monitoring of the changes is passive and thus avoids potential health and/or environmental concerns related to subjecting the body to electrical signals and/or radiation. Furthermore, the changes in the ear uniquely correspond to one of a variety of initiating actions (*e.g.*, have unique signal signatures) which allows a plurality of different initiating actions to be detected and distinguished for a system requiring multiple control functions.

According to one aspect of the present invention, the sensor is in electrical communication with processing circuitry and senses changes of the ear and converts the sensed changes into an electrical signal. The electrical signal is then processed to provide a control instruction or some other useful output, such as an output similar to that produced by a computer mouse or joystick which corresponds to the particular initiating action. Exemplary control functions may include, but are not limited to: controlling a video game display, controlling a piece of medical equipment such as a wheelchair, and controlling computer functions to effectuate a handless mouse, keyboard, joystick, etc.

Turning now to the figures, FIG. 1A is a block level diagram which illustrates a system 10 for detecting initiating actions and providing an output which corresponds to the detected action. The system 10 includes a pressure sensor 12 coupled to processing circuitry 14. The sensor 12 includes an internal ear microphone as described in more detail below. The sensor 12 detects changes within the ear that result from an initiating action by

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the person wearing the sensor 12. The processing circuitry 14 includes a data acquisition circuit 16, such as a computer sound card. Other example data acquisition circuits include a PCI9118HG data acquisition card manufactured by Adlink Technology or a DAQi250 data acquisition card manufactured by Ines Company Gmbh in Germany. The data acquisition circuit can include an analog to digital (A/D) converter for converting an analog signal into digital signal data. A buffer 17 can be provided to store portions of signals from data acquisition circuit 16 (as those signals may be relatively continuous) until a filtration and signal detection can be carried out as are described further below. The processing circuitry 14 also includes a processor 18 for receiving the digital signal data from the data acquisition circuit 16 and buffer 17 and performing various signal processing functions on the digital signal data to detect the occurrence of an initiating action and determine the type of initiating action. The system 10 also includes an output peripheral 20 coupled to the processor 18 for executing one or more control instructions provided by the processor 18 which correspond to the initiating action. One skilled in the art will recognize that, depending on the application, the analog to digital converter function and the buffering function can be omitted or performed by software.

FIG. 1B is a block level diagram which illustrates another embodiment of a system 22 for detecting initiating actions and providing an output which corresponds to the detected action. The system 22 includes the sensor 12 having an internal microphone (described in greater detail below). The electrical signal produced by the sensor 12 is optionally amplified with an amplifier 24. However, in some applications, the amplifier 24 is not needed depending on the equipment receiving and processing the electrical signal output by the sensor 12.

The electrical signal is then received by an interface adapter 26. The interface adapter 26 can be a mechanical connection such as an electrical connector to be received by a corresponding jack on the device which ultimately receives and processes the electrical signal output by the sensor 12. The interface adapter 26 can also include more complex instrumentation, such as a universal serial bus (USB) adapter. As one skilled in the art will appreciate, depending on the processor, device or application receiving the electrical signal produced by the sensor 12, the interface adapter 26 is optional or can be any device for establishing physical and/or logical connection between the sensor 12 and the device to receive the electrical signal produced by the sensor 12.

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The electrical signal is then received by a computer interface 28. The computer interface 28, as with the amplifier 24 and the interface adapter 26, is optional depending on the configuration of the system 22. The computer interface 28 can be a mechanical connection such as a jack or receptacle, or can be more complex instrumentation, such as a sound board for a computer. In one embodiment, the amplifier 24 and/or the interface adapter 26 are omitted and the signal produced by the sensor 12 is input to processing circuitry 14' via a conventional computer sound board, the sound board being used to convert the analog signal produced by the sensor 12 into a digital signal.

The electrical signal is then processed by processing circuitry 14'. The processing circuitry 14' can take on the form of the processing circuitry 14 illustrated in FIG. 1A and can optionally include such devices such as an analog to digital converter, a buffer, and/or a processor 18'. The processor 18' can be configured to execute logic for a variety of applications that make use of the signal generated by the sensor 12.

In the illustrated embodiment, the processor 18' is programmed to execute a voice recognition software program. As is known in the art, voice recognition software converts spoken language into text which can be electronically manipulated such as with a word processing program. The text can be output to an output peripheral 20. For example, the text can be displayed on a display or printed by a printer. The text can also be stored for later retrieval by a storage device 30 on a computer-readable medium including, but not limited to, magnetic and optical storage devices. Alternatively, the electrical signal itself or recognized speech derived therefrom can be used to control machinery such as a self-powered wheelchair, a robotic arm, a computer, transportation vehicles, domestic and industrial appliances or machinery, and the like.

FIG. 1C is a block level diagram which illustrates a system 32 for detecting initiating actions and providing an output which corresponds to the detected action. The system 32 includes the sensor 12 having at least an internal microphone 34. The sensor 12 and internal microphone 24 will be described in more detail below. The sensor 12 outputs an electrical signal which is optionally amplified by amplifier 24. Next, the electrical signal is received by a communication device 36 via an appropriate interface adapter 26, if needed. The communications device can be, for example, a telephone which is either hard-wired to a telephone system or, as shown in the illustrated embodiment, is a wireless telephone such as a cellular phone. Alternatively, the communications device 36 can be a

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radio, portable computing device, or the like. In one embodiment, the signal generated by the sensor 12 is input into a personal computer, personal digital assistant (PDA) or the like and is transmitted to another location or device via a network (e.g., a LAN, a WAN, the Internet, etc.). In another arrangements, the communications device can be replaced with other types of devices such as a recorder for recording the user's speech for later retrieval or an amplification system for broadcasting the speech to a live audience.

In addition, the sensor 12 can include a speaker 38 used to transmit (i.e., broadcast) sounds to the user. In this way, the sensor 12 and communication device 34 can be used as a bi-directional communication apparatus.

FIG. 1D is a block level diagram which illustrates a system 40 for detecting initiating actions and providing an output which corresponds to the detected action. The system 40 includes the sensor 12, which preferably includes the internal microphone 34 (FIG. 2B) described in more detail below. The sensor 12 is connected to a medical diagnostics device 42 via, if desired, an amplifier 24, interface adapter 26 and computer interface 28. The medical diagnostics device 42 has an audio output 44 (e.g., speaker, stethoscope attachment, etc.) so that detections of air pressure by the sensor 12 corresponding to biological processes (heart beat, breath sounds, gastronomic sounds, etc.) of a patient can be heard and studied by medical personnel. To help the medical personnel listen to the biological processes, the medical diagnostics device can optionally include filters (in hardware or software) to isolate these sounds before broadcasting the sounds to the medical personnel. In another embodiment, the medical diagnostics device 42 has a video output 46 (e.g., an LCD or CRT screen) to display a video representation of the detected air pressure changes. For instance, the sound generated by the heart within the patient's ear and detected by the sensor 12 can be converted into a graph (e.g., amplitude versus time) and displayed on the video output 46 in similar fashion to display of an electrocardiogram.

The medical diagnostics device 42 can also include a recording device 48 (e.g., a digital or analog data storage medium) to store detections of the patient's biological processes. The recorded sounds can be compared to subsequent detections. This comparison can be made by a person or by software programmed to isolate differences between the first and second detections.

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The medical diagnostics device 42 can also include a processor 50 which executes programmed logic (e.g., software) to analyze the detected air pressure changes. The software can be programmed to search the signal produced by the sensor for patterns, irregularities (including those normally undetectable by a human), differences from a baseline detection and so forth.

In addition, data acquired by the medical diagnostics device 42 from the sensor 12, or data calculated by the medical diagnostics device 42, can be transmitted to other locations. For example, the medical diagnostics device 42 can be placed in an ambulance and transmit information about a patient to a destination hospital. As an alternative example, the data could be gathered by one medical worker and transmitted to another medical worker, such as a specialist located in another city.

In another embodiment, the sensor can be located remotely from the medical diagnostic device 42. For example, a person could use the sensor 12 from a location such as his/her home. The information detected by the sensor 12 could then be transmitted to a remote medical diagnostics device 42. The transmission could be carried out over a telephone or other communications device as discussed above with respect to the system 32 illustrated in FIG. 1C. Alternatively, the signal generated by the sensor 12 can be input to a personal computer, a PDA or the like and transmitted to the medical diagnostics device 42 via a network (e.g., a LAN, a WAN, the Internet, etc.).

FIG. 1E is a block diagram which illustrates a system 52 for detecting initiating actions and providing an output which corresponds to the detected action. The system 52 includes the sensor 12, preferably having the internal sensor 34. In the illustrated embodiment of the sensor 12, the sensor 12 has an additional pressure detector, or external microphone 54. The use of the external microphone 54 is optional. As will be described in more detail with respect to FIG. 2B below, the internal microphone 34 is disposed on a housing 150 (FIG. 2B) to be inserted at least part way into the ear of a person. That is, the internal microphone 34 is insert at least into a cavity defined by an external portion of the ear (described below in more detail with respect to FIG. 2A). As a result, the internal microphone 34 detects air pressure changes located within the ear. These air pressure changes are the result of initiating actions and, in part, to externally generated sound, such as other people's speech, environmental noise and the like. The external microphone 54 is disposed on the housing 150 (FIG. 2B) and generally points away from the person wearing

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the sensor 12. As a result, the external microphone 54 detects air pressure changes located outside the ear. These air pressure changes are the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to initiating actions of the person wearing the sensor 12 and particularly the speech of that person.

The internal microphone 34 is connected to processing circuitry 14" or another device used to receive the electrical signals respectively produced by the microphones 34 and 54 in response to sensed air pressure changes. Each microphone 34 and 54 is connected to the processing circuitry 14" using separate and distinct inputs, or channels. In one embodiment, the internal microphone 34 is connected to the left channel input of a computer sound card, or computer interface 28a, via an amplifier 24a and an interface adapter 26a. The external microphone 54 is connected to the right channel input of the computer sound card, or computer interface 28b, via an amplifier 24b and an interface adapter 26b.

The processing of the unique signals generated by the microphones 34 and 54 will be described in more detail below with respect to FIG. 3B. The unique signals can be used to distinguish between user produced initiating actions and externally produced sounds. In alternative configurations of the system 52, the system 52 can be adapted to cancel externally generated noise from the detections of the internal microphone 34.

Testing has shown that when the user speaks (or produces an air pressure change in the ear from another type of initiating action), the strength of the signal generated by the internal microphone 34 corresponding to the speech (or initiating action) is usually stronger than the strength of the signal generated by the external microphone 54 that corresponds to the same speech (or initiating action). Testing has also shown that when a sound is generated by a source other than the user (e.g., another person speaking), the strength of the signal generated by the external microphone 54 corresponding to the sound is usually stronger than the strength of the signal generated by the internal microphone 34 that corresponds to the sound. These sound energy differentials, or relative signal strengths, can be used to determine whether the source of the detected air pressure change is from the user or external to the user.

For instance, if the user is speaking, the internal microphone 34 may detect the user's speech and external noise (e.g., the hum of an air conditioner and a computer). In this example, the external microphone 54 may detect the user's speech, but to a smaller

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degree than the internal microphone 34. The external microphone 54 will also detect external noise and that external noise will be detected to a larger degree by the external microphone 54 than by the internal microphone 34. Accordingly, the processing circuitry 14" can be programmed to compare the signals and discern that the source of the speech is from the user. In addition, the processing circuitry 14" can be programmed to isolate the user's speech by comparing the two signals to identify the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be used in connection with voice recognition software as indicated above. In alternative embodiments, the foregoing technique can be used to distinguish between the user's speech and another individual's speech. Both person's speech can then be analyzed as desired. As one skilled in the art will appreciate, the foregoing technique can be applied to the detection of other type of initiating actions, including thoughts, movements related to the oral cavity, biological processes and the like.

Referring to FIGs. 2A and 2B, an external view and a cross-sectional view of an ear 100 are respectively illustrated. FIG. 2B also shows the sensor 12 in cross-section. According to Henry Gray's famous text "Anatomy", the human ear is divided into three parts, including the external ear 102, the middle ear (or tympanum) 104 and the internal ear (or labyrinth) 106. The middle ear 104 and the internal ear 106 will not be described in great detail herein. The external ear 102 includes an expanded portion, or a pinna 108 (also referred to as an auricle), and an ear canal 110 (also referred to as a meatus or auditory canal). The pinna 108 serves to collect vibrations of the air surrounding the person's head. The ear canal 110 conducts those vibrations to the tympanum, or ear drum 112.

The pinna 108 has a generally ovoid form with a larger end directed upward and having an outer surface that is irregularly concave and directed slightly forward. The pinna 108 has a number of eminences and depressions. Typically, the ear 100 has a prominent and curved rim, or helix 114. Generally parallel to the helix 114 is another curved prominence, or antihelix 116. The antihelix 116 bifrucates to form a triangular depression, or a fossa of the antihelix 118 (also referred to as a fossa triangularis). A narrow, curved depression located between the helix 114 and antihelix 116 is referred to as fossa of the helix, or scapha 120. The antihelix 116 also curves around a deep, capacious cavity, or the concha 122 (the concha 122 being divided by the commencement of the helix 114, or crus helicis, into an upper part, termed the cymba conchae, and a lower part, termed the cavum

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conchae). The concha 122 leads inward to an opening of the ear canal 110. In front of the concha 122 and projecting backward (usually over the opening of the ear canal 110) is a pointed eminence, or tragus 124. Opposite the tragus 124 is a tubercle, or antitragus 126. A notch-like depression, or incisura intertragica 128, is disposed between the tragus 124 and antitragus 126. A lobule 130 is present under the tragus 124 and antitragus 126.

The ear canal 110 is an oval cylindrical passage extending from a bottom of the concha 122 to the ear drum 112. The ear canal 110 is about an inch and a half in length when measured from the tragus 124 to the ear drum 112. When measured from the bottom of the concha 122 to the ear drum 112, the ear canal is about an inch long. The ear canal 110 forms a gradual "S-shaped" curve and is directed, at first, inward, forward and slightly upward (i.e., pars externa). The ear canal 110 then passes inward and backward (i.e., pars media) and then passes inward, forward and slightly downward (i.e., pars interna).

It is not certain what physical, chemical or neural mechanism causes or generates the changes in air pressure in or near the ear or sounds to come from the ear in response to various initiating actions of the user. However, due to the connection of the oral cavity to the ear via the eustachian tube, movements and speech may cause an air pressure, a change in air pressure or an air flow to or from the ear leading to a detectable air pressure which can be detected by the sensor 12. Regardless of the exact physical, chemical or neural mechanism, empirical testing has confirmed that the initiating actions described herein generate pressure changes in or near the ear of the person and that the air pressure changes have substantially their own signature and are thus substantially unique for a given initiating action of an individual. Consequently, the air pressure changes can be monitored near the ear and used to detect the initiating action of a user.

In addition, some initiating actions, such as movements of the tongue, can produce a detectable pressure wave that has a strength corresponding to the direction, speed and/or intensity of the movement. Although the invention may use a single sensor 12 in one ear 100, two sensors 12 (one in each ear 100) can be used to obtain additional information whereby each sensor 12 detects different information and/or cumulative information, which can be correlated by the processor.

The present invention uses various forms of the terms "changes in air pressure", "changes within the ear" and "sounds emanating or coming from the ear" in their broadest sense to characterize the parameter being measured. Changes in air pressure may

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alternatively be characterized as sound waves. These sound waves (or vibrations) may propagate through mediums other than air, such as bone and tissue. As is well known by those skilled in the art, as a sound wave spreads out from its source its intensity falls off (the energy per unit area decreases with the inverse square of the distance), but the total energy is constant. Consequently, it is desirable to have the microphone 34 sufficiently close to the source of the sound wave so that the intensity level of the detected changes will be as large as possible.

The frequency range at which sound waves are audible by humans is about 20 Hz to about 20 KHz, however, the present invention is not concerned with whether the changes within the ear are audible since the microphone 34 is selected to be sufficiently sensitive and to have a frequency detection range which is sufficient to detect air pressure changes at high or low frequencies. Therefore, any frequency range may be monitored which is suitable to detect initiating action (e.g., about 10 Hz to about 20 KHz) and such variations are contemplated as falling within the scope of the present invention.

FIG. 2B illustrates the sensor 12 inserted at least partially into the ear 100 of a person (i.e., at least within the cavity defined by the pinna 108, if not deeper within the ear 100 such as within the concha 122, at the opening of the ear canal 110 or slightly into the ear canal 110). The sensor 12 includes a housing 150, the internal microphone 34 and, optionally, an external microphone 54. In the illustrated embodiment the housing 150 has an exterior portion 152 and an interior portion 154. The housing 150, however, can take on a number of different configurations. The housing preferably resembles the housing of a hearing aid and particularly a digital hearing aid for similar insertion, or partial insertion, into the ear 100 and/or ear canal 110. The exterior portion 152 is optional. The illustrated external portion 152 is made from a material such as plastic and is wider than an opening of the ear canal 110 to engage the pinna 108 or, alternatively, to engage the ear 100 to cover the whole pinna 108 (e.g., similar to that of a common headphone assembly). In one embodiment, the external portion 152 fits within the concha 122 and is retained, at least in part, by the tragus 124 and/or the antitragus 126. Such arrangement at least partially insulates the interior microphone 34 from externally generated noise and air pressure changes.

Alternatively, the housing 150 can resemble a miniature ear phone as found in conventional wireless telephone headsets or as used with personal audio/music players.

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The earset 150 can be retained by insertion into the ear 100, by a member disposed over or hanging from the ear and/or by a headset assembly. The illustrated housing 150 is made from any suitable material, such as plastic, rubber or a gel-like material. Optionally, the housing 150', or portions thereof, can be made of pliable material, sound absorbing (or sound proofing) material and/or include sound insulating material such as foam.

The interior portion 154 can be made from a pliable material, such as rubber, and is sized and shaped to snugly fit within the ear 100. Such material is sometimes referred to in the art as being "soft-solid." Optionally, the entire housing 150, or portions thereof, can be made of pliable material, sound absorbing (or sound proofing) material and/or include sound insulating material such as foam.

The internal microphone 34 is disposed on the interior portion 154 of the housing 150. Accordingly, the interior portion 154 is sized and shaped to position the internal microphone 34 with respect to the ear 100 and/or ear canal 110 as desired. On one embodiment, the interior portion 154 positions the internal microphone 34 in the concha 122. In another embodiment, the interior portion 154 positions the internal microphone 34 at the opening of the ear canal 110 where the ear canal 110 meets the concha 122. It should be appreciated that the interior portion 154 and/or internal microphone 34 need not enter the ear canal 110. In yet another embodiment, and as illustrated by example in FIG. 2B, the interior portion 154 positions the internal microphone 34 within the ear canal 110. In another embodiment, the interior portion 154 positions the microphone 34 and/or a speaker 38 (FIG. 16, described below) in the concha 122. In another embodiment, the interior portion 154 positions the microphone 34 and/or the speaker 38 at the opening of the ear canal 110 where the ear canal 110 meets the concha 122. It should be appreciated that the interior portion 154, microphone 34 and/or the speaker 38 need not enter the ear canal 110. In yet another embodiment, the interior portion 154 extends into the ear canal 110 and positions the microphone 34 and/or the speaker 38 within the ear canal 110. When positioned within the ear canal 110, the internal microphone 34 enters the ear canal 110 at a depth of about one tenth of an inch to about a half of an inch measured from the opening of the ear canal 110.

The housing 150, including both the exterior portion 152 and the interior portion 154, can be custom designed for the individual to form a close and comfortable fit with the ear of the individual. Alternatively, the housing can have a standard, or "stock", design for

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all individuals which is fabricated in a number of sizes. As one skilled in the art will appreciate, many alternative configurations for the housing 150 are possible and each are considered to fall within the scope of the present invention.

In one embodiment, the microphone 34 is held in place with an adhesive. The interior portion 154 of the housing 150 has a recess 156 in which the microphone 34 is placed. Preferably, the air adjacent the ear drum 112 and/or the air in the ear canal 110 will be in fluid communication with the air adjacent the microphone 34. In this way, sounds or air pressure changes generated within the ear 100 will be transmitted directly to the microphone 34 through an air medium. Other sounds conducted by the bones and tissue of the user may be transmitted to the microphone 34 mechanically via the housing 150. The microphone 34 in one embodiment is about one mm to about six mm in diameter. One skilled in the art, however, will appreciate that the microphone 34 is not necessarily round. In an alternative embodiment, the microphone 34 is disposed within a hollow cavity defined by the housing 150 and an opening in a wall of the internal portion 154 of the housing 150 is provided to establish fluid communication of the air inside the housing 150 and the air inside the ear canal 110.

By inserting the microphone 34 in the ear 100 and/or ear canal 110, the microphone 34 is shielded from environmental noise. More specifically, the housing 150 and the head of the user at least partially blocks externally generated sound waves before reaching the microphone 34.

In one embodiment, the microphone 34 is a directional microphone adapted to detect air pressure changes from a particular direction. The directional microphone is aimed as desired to detect various different air pressure changes generated by different type of initiating actions. For example, the microphone can be aimed towards a desired source, such as the ear drum 112, the ear canal 110 wall, or the bone structure surrounding the opening of the ear canal 110. Since the microphone is directional, the microphone will detect less environmental noise (the source of that noise is outside the ear and from a direction which does not typically coincide with the directional sensitivity of the microphone). In other embodiments of the present invention, the microphone 34 is a noise canceling microphone to further reduce the detection of environmental noise.

It should be appreciated that the microphone 34 can be moved closer to or further away from various anatomical structures within the ear 100 as desired for factors such as

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comfort, detection of certain air pressure changes caused by a specific type of initiating action, to optimize detection of the user's speech, and the like. It is noted that the microphone 34 is capable of detecting speech of the user speaking at a variety of volumes.

The microphones 34 and 54 can take on a number of different forms, such as a silicon or semiconductor microphone (also referred to as a pressure sensor) such as the devices disclosed in U.S. Patent Nos. 4,533,795, 4,922,471, and 6,156,585, each of which are incorporated herein by reference in their entireties. It is noted that the foregoing devices are exemplary and other electroacoustic transducers from a number of different vendors can be used.

In the illustrated embodiment, when the sensor 12 is inserted into the ear 100, a slight positive pressure is created by compression of the air within the ear canal 110. It is believed that this slight positive pressure, which is undetectable to the individual, improves the overall operating characteristics and sensitivity of the sensor 12.

The microphone 34 is capable of detecting various forms of initiating action, including, for example, the speech of a person speaking at a variety of volumes, grunts, whistles, singing, coughs, physical movements of the person such as clicking the person's tongue against the roof of the person's mouth, opening the person's jaw, the person's breathing, the person's heartbeat, etc. In addition, pressure changes in the ear caused by thoughts can be discerned by detecting the changes within the ear when a trained individual thinks a thought.

It has been shown that the use of one sensor 12 with only the internal microphone 34 (i.e., without an external microphone 54) in one of the ears 100 of an individual can be used to detect the initiating actions described herein. However, empirical testing has shown that by using two such sensors 12, one in each of the ears of an individual, the performance of systems such as the systems 10, 22, 32, 40 and 52 improves by about six times due to the detection of more information. It is noted that in certain circumstances, the use of a sensor 12 in both ears is discouraged (such as when using a cellular telephone while driving a car). It is noted that the circuitry, processing and logic routines described herein can be modified to process signals from a second earset 12 or sensor 12 (the sensors 12 with or without an external microphone 54). Such modifications will be apparent to one of ordinary skill in the art.

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Although not illustrated in FIG. 2B, a speaker 38 (FIGs. 1C and 16) can be placed on or within the housing 150 of the sensor 12 to transmit sounds to the individual. For example, when using the sensor 150 as a microphone for a communications device 36, the sensor 12 can also be used to broadcast received sounds to the user.

As indicated, the sensor 12 can also include the external microphone 54. The external microphone 54 is disposed on the exterior portion 152 of the housing 150 and is pointed away from the wearer of the sensor 12 assembly. The external microphone 54 can be secured to the housing 150 with adhesive and can be disposed in a recesses defined by the exterior portion 152. Similar to the internal microphone 34, the external microphone 54 is small (e.g., about one mm to about six mm, but need not be round).

The position of the external microphone 54 allows the external microphone 54 to detect air pressure changes present near the wearer's ear but have a source external to the wearer of the sensor 12. In one embodiment, the external microphone 54 can optionally be a noise canceling microphone as is known in the art.

The internal microphone 34 is electrically coupled to the amplifier 24 with a conductor 158 and the exterior microphone 54 is electrically coupled to the amplifier 24 with a conductor 160. Although illustrated as a single unit in FIG. 2B, the amplifier 24 contains separate amplifying circuitry 24a and 24b (FIG. 1E) for each signal respectively generated by the internal microphone 34 and the external microphone 54. The amplifier 24 is electrically coupled to the interface adapter 26 (not shown in FIG. 2B) with a conductor 162. More specifically, the conductor 162 carries the amplified signals from the microphones 34 and 54 separately to separate interface adapters 26a and 26b. As one skilled in the art will appreciate, if the amplifier 24 is omitted and the microphones 34 and 54 and/or speaker 38 can be coupled directly to the interface adapter(s) 26 by way of a conductor or conductors. It is also noted that if two sensors 12 are used by placing one sensor 12 in each ear 100 of a person, the processing circuitry or other device receiving signals from the sensors 12 will receive four signals corresponding respectively to the internal microphone 34 and the external microphone 54 of each sensor 12.

The microphones 34 and 54 of the sensor 12 monitor the changes in air pressure and convert the pressure changes to an analog electrical signal 190, as illustrated in FIG. 2C. In the signal 190 there are at least two signal components, a high frequency component 192 and a low frequency component 194. In addition, other frequencies may also exist

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within the electrical signal 190 and the present invention can be adapted to analyze the various signal frequencies, as described herein.

In an alternative embodiment, the internal microphone 34 and the external microphone 54 are mounted to a headphone style assembly having a band disposed around the user's head and hubs disposed over one or both ears 100 of the user. One exemplary headphone and internal microphone assembly which may be used is the Model 4190 microphone manufactured by Brüel & Kjær in Denmark.

A method 200 for carrying out the present invention is disclosed in FIG. 3A. The method 200 includes detecting an initiating action by monitoring a change in the ear 100 at step 202 with the internal microphone 34 of the sensor 12. The external microphone 54 is optional in carrying out the method 200. Once the initiating action is detected at step 202, one or more control instructions or other output(s) which correspond to the detected initiating action is provided at step 204 to effectuate the desired output function.

A method 200' for carrying out the present invention is disclosed in FIG. 3B. The method 200' includes the detection of changes in air pressure with both the internal microphone 34 and the external microphone 54. More specifically, in step 202', changes in air pressure within the ear 100 and/or ear canal 110 are detected by the internal microphone 34. These detections can be the result of an initiating action, but can also include a sound component generated external to the person wearing the sensor 12 and/or a sound component generated by speech of the wearer and traveling from the wearer's mouth, through the air and to the wearer's ear 100 area. Also in step 202', changes in air pressure near, but outside, the ear 100 of the person wearing the sensor 12 are detected by the external microphone 54. These detections are mainly the result of sounds generated external to the person wearing the sensor 12 and/or a sound component generated by speech of the wearer and traveling from the wearer's mouth, through the air and to the wearer's ear 100 area. The detections are converted by the microphones 34 and 54 into respective electrical signals, herein referred to as an internal signal and an external signal.

Next, in step 206, the internal signal and the external signal are compared. When the user speaks or produces an air pressure change in the ear 100 from another type of initiating action, the strength of the signal generated by the internal microphone 34 corresponding to the initiating action is usually stronger than the strength of the corresponding signal generated by the external microphone 54. Testing has also shown that when a sound is

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generated by a source other than the user (e.g., another person speaking), the strength of the signal generated by the external microphone 54 corresponding to the sound is usually stronger than the strength of the corresponding signal generated by the internal microphone 34.

Using the signal strength differential, the source of the detected air pressure changes (i.e., from the user or external to the user) can be reliably determined. From further comparison of the internal signal and the external signal, two components from each signal are derived. The two components for each signal include a component relating to air pressure changes caused by the initiating action and a component relating to externally generated noise and/or sound, which may or may not include speech of the user detected by the external microphone 54 depending on the system in which the sensor 12 is being used. It is noted that for initiating actions other than speech, the initiating action component of the external signal may be nearly undetectable.

Subsequent processing of the internal signal and external signal will depend on the system in which the sensor 12 is being used and which types of initiating actions are being detecting. For example, the signals can be used by the processor or other device receiving the signals to invoke a corresponding output signal or signals. However, in most applications, it will be advantageous to isolate the component of the internal signal generated by the initiating action and, if desired, eliminate to the fullest extent possible any externally generated component of the internal signal. Therefore, in step 208, the method 200' distinguishes the initiating action component of the internal microphone signal. In another embodiment of step 208, the method 200' can cancel noise (or all sounds) detected by the external microphone 54 from the detections of the internal microphone 34. More specifically, the externally generated component of internal signal is filtered from or mathematically removed from the internal signal. The resulting signal from step 208 is used in step 204' to provide the control instruction or output response appropriate for the initiating action.

Similar processing to process the external signal can be carried out in step 210 to derive a signal of externally generated sounds, such as a to isolate another person's speech from noise and from speech of a person wearing the sensor 12. Thereafter, in step 212, a control instruction or output response appropriate for the externally generated sound can be produced.

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A method of detecting changes in the ear 100 (step 202) is illustrated in FIG. 4. The microphone 34 is placed in the ear 100 of the user who's initiating actions are to be detected at step 214. However, initiating actions may result in changes at or near other parts of the body. Therefore, it is contemplated that in alternative embodiments of the present invention, the sensor 12 may be located on, in or near other parts of the body and any detection of initiating actions by analyzing physiological or air pressure changes is contemplated as falling within the scope of the present invention.

The changes in the ear 100 are monitored with the microphone 34 at step 216 and are converted to an electrical signal at step 218 for subsequent analysis. After conversion into an electrical signal at step 218, the electrical signal is analyzed to detect the initiating action at step 220. Although it is conceivable that the initiating action may be detected at step 220 simply by analyzing the signal corresponding to changes in air pressure or changes in the ear 100 without additional data processing, step 220 typically includes data processing in conjunction with the signal analysis. It is noted that when a sensor is used which has an external microphone 54, then the foregoing logic can be modified to include the capture of an electrical signal corresponding to air pressure changes outside the ear 100 and using that signal in the analysis of step 220 as described above.

An exemplary method of analyzing and processing the electrical signal which corresponds to the monitored pressure (step 220) is illustrated in FIG. 5. Again, the analyzing logic can be adapted as described herein to include processing of a second electrical signal derived from an external microphone 54. The analog electrical signal, an example of which is illustrated in FIG. 2C, is converted into a digital signal at step 222, as illustrated by example in FIG. 6. As is well known by those skilled in the art, an analog signal may be converted into a digital signal by sampling the analog signal at a selected frequency and identifying the signal amplitude at each sampling point. Each sampled data point is then saved as a digital word in a memory and used for further analysis. In FIG. 6, a sampled analog signal is illustrated in which the dotted line illustrates the exemplary analog signal for a particular time period and the plurality of points on the dotted line represent sampled amplitude values which are saved in the memory. It is desirable that the sampling frequency be sufficient to capture enough data points to adequately represent the analog signal. As an example, the sampling rate of the may be 32 KHz and the total signal time length to be analyzed may be 2048 mSec. Alternatively, however, other sampling rates

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and data acquisition time frames may be utilized and such variations are contemplated as falling within the scope of the present invention.

Once the analog signal has been converted into digital signal data at step 222, the digital data is analyzed and processed by, for example, a signal processor to detect the presence of an initiating action at step 224. As indicated above, step 224 may include comparison of the internal and external signals, and filtering or mathematical manipulation of the signals based on the comparison to isolate a desired component(s) of the detections and/or to remove an undesired component(s) of the detections. In addition, or as an alternative, the analysis and processing of the data may be performed in a plurality of segments, for example, as illustrated in FIGS. 7 and 8. As illustrated in FIG. 7, the processing of step 224 can include analysis of a first data segment in step 226, followed by the analysis of a second data segment at step 228. Once various data segments have been analyzed separately, the data segments are analyzed together at step 230. If all the data segments have not yet been analyzed at step 232, the method returns to step 228 and the next data segment is analyzed, after which all previous segments are then analyzed together at step 230. The process continues until all the data segments have been analyzed at step 232, thus allowing a conclusion to be generated using the analyzed data segments at step 234.

The data segment analysis may be seen graphically in FIG. 8, wherein digital signal data 236 is illustrated as being continuous for the sake of simplicity. The total length of data for analysis may be separated into 64 segments that are each 32 mSec in length. Note that the signal 236 contains both a high frequency component 238 and a low frequency component 240. Since data relating to an initiating action potentially may be found in either component or the initiating action data may span multiple data segments, the data segments may be analyzed separately as well as together. Thus, at step 226 of FIG. 7, the first data segment is analyzed (region 242), at step 228 the second data segment is then analyzed (region 244) and at step 230 both data segments are analyzed together (region 246). The process then continues for all the data segments; consequently, the data analysis of the present invention may analyze both the high frequency and low frequency signals to detect the initiating action.

Returning back to FIG. 5, once the data considered to be initiatory data has been found in the data signal from the sensor at step 224, subsequent analysis is performed to

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determine the type of initiating action at step 248. Such analysis can include feature extraction and analysis using a neural net trained to recognize respective signals/features of such signals. If the neural net determines that the signal represents an initiating action of a particular character, then it provides an output or effects the production of an output representative thereof for use as described herein. Alternatively, however, other techniques may be utilized to identify the initiating action type once the initiating action is detected and any such technique is contemplated as falling within the scope of the present invention.

One exemplary method of analyzing the digital signal in data segments (step 224) is illustrated in FIG. 9. For each data segment of 32 mSec, the data is converted from the time domain to the frequency domain at step 250 using, for example, a Fast Fourier Transform (FFT) as is well known by those skilled in the art. As is well known, a time domain signal f(t) is linked with the frequency domain f(jω) according to the following equation:

$$F(f(t)) = \int f(t)e^{-j\omega t}dt = f(j\omega) \qquad \text{Eq. 1}$$

wherein F(f(t)) is a traditional Fourier transform. As is well known by those skilled in the art, a Fast Fourier Transform is related to the traditional Fourier transform since the Fast Fourier Transform is an efficient algorithm for computing discrete Fourier transforms. After the digital signal data is converted into the frequency domain via the Fast Fourier Transform, the frequency domain data is processed to distinguish data relating to initiating actions from noise data at step 252. As is well known by those skilled in the art, the separation of data from noise is often simplified in the frequency domain because unlike noise, the data signal has some physical characteristics. Though the data signal in the time domain has an amplitude which is less than the noise, the data signal has a greater amplitude than the noise in the frequency domain. Therefore the Fast Fourier Transform is a typical method for noise separation. In addition, or as an alternative, signal data from the external microphone 54 can be used to derive signal components as described in greater detail above.

The details surrounding the data processing of the digital signal data may be accomplished through a variety of data processing techniques as is well known by those skilled in the art and any data processing methodology is contemplated as falling within the scope of the present invention. Although many different data processing methodologies

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may be employed, an example methodology is disclosed below in conjunction with the following method and system.

Turning to FIG. 10, a system 270 and method are depicted. One or more sensors 12 sense changes in the ear(s) 100 of the user and, if desired, external to the ears(s) 100 of the user. The analog signal(s) from the sensor(s) is provided via a data acquisition card, analog to digital converter, and buffer 16/17 to the processor 18. The signal received from the sensors is substantially continuous at least over respective time periods while the initiating action is being performed; and, therefore, the buffer 17 is able to store sufficient portions of the signal for analysis by the processor 18.

In one embodiment, the processor 18 can include a band pass filter 272, power spectrum filter 274, signal detector 276, signal comparator 278, feature extraction portion 280, and neural network 282. The system 270 also includes an output 284 which may be coupled to a desired output peripheral 20.

Summarizing operation of the overall system 270, the sensor 12 and data acquisition, A/D, and buffer 16/17 supply a signal to the band pass filter 272. An example of a raw data or raw signal supplied to the band pass filter 272 is illustrated at 286 in the graph of FIG. 11. The raw signal is presented to the band pass filter and is operated on by the band pass filter in the frequency domain using fast Fourier transform techniques. The fast Fourier transform transforms the digital signal to the frequency domain. In the band pass filter frequencies that are above or below the frequency range representing initiating action, for example, are removed from the signal. The frequencies which are above and below the frequency range of interest typically are due to noise. An example of the signal after being filtered by band pass filtration is illustrated at 288 in the graph of FIG. 12. Various band pass filters may be used to provide the band pass filtration to strip from the signal frequencies above and below the desired frequency range.

Portions of the signal represented at 288, such as portions 288a and 288b may not be representative of initiating action, e.g., since they are relatively low power and, therefore, still may be noise. Therefore, the power spectrum filter 274 does a further filtration according to power of the various portions of the signal to determine portions which are due to initiating action and, accordingly, should be considered for further analysis. The power filtration may be carried out by slicing the signal of FIG. 12 to multiple windows or segments, such as those represented at 290a, 290b, ... 290i in FIG. 13.

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It will be appreciated that the graph of FIGS. 12 and 13 are of different scales. Thus, the windows 290a-290i slice a time frame between times t_a and t_b, which is the same time frame between times ta and tb illustrated for the signal portion 288c in FIG. 12.

Power filtration suppresses or eliminates those frequencies where the power of the signal does not exceed a prescribed threshold or value and, thus, likely represent noise. By eliminating those noise components the signal representing the initiating action can be found. To perform the power filtration, for example, a power spectrum for the signal is calculated using the following formula:

$$P_f = Mag_f^2 + Phase_f^2 \qquad \text{Eq. 2}$$

where $f=F_0...F_i$, and $[F_0; F_i]$ represent the frequency range in which the signal resulting from tongue action is expected.

In words, the power is a function of the magnitude squared plus the phase squared. The power as calculated above for respective windows 290a, 290b...290i is reconstructed to make a graph 292 shown in FIG. 14. The graph 292 shows values 292a, 292b...292i from the mentioned computation. The curve 292 is relatively smooth and it represents power spectrum where a signal caused by initiating action may be found in the graph 288 of FIG. 12. Usually the signal caused by initiating action is of relatively high power and occurs in time approximately at the middle of the graph 292, e.g., over the time frame T shown in FIG. 13. It is the time at which this time frame occurs and the duration of it that detects the signal or locates when in time the signal 288 represents the initiating action. Therefore, the signal detector component 276 of the system 270 in effect is the combination of the graphs of FIGS. 12 and 13 and the analysis just described to provide the signal detection function. The time frame T also is shown in the graph of FIG. 12 and, thus designates that portion 288d (in the illustrated blocked window 288e) of the signal 288c which is to be considered produced by the initiating action.

As an alternative to, or in addition to, the filtration and detection operations of the power spectrum filter 274, the signal detector 276 and the feature extraction 278, the signal comparator 278 can carry out method 200' to produce a signal indicative of the initiating action, or signal portion 288d. The signal portion 288d or at least a feature of that signal then may be supplied to the neural network 282. Using feature extraction principles a feature of the signal portion 288d is supplied to the neural network. In the neural network 282 the feature is compared with other features on which the neural network has been

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trained. Training of neural networks is a well known procedure. The neural network can determine whether an input feature, such as that representative of the signal portion 288d, is an acceptable match or correlates acceptably to the training of the neural network; and, if affirmative, then the neural network causes production of the output 284. The output 284 may be an electrical signal which is analogous to the type of signal produced by a computer mouse, joystick or other device. Thus, in one example, a tongue action whereby the tongue of the user is moved from the front bottom left of the mouth to the back top right of the mouth may cause the production of an output signal analogous to a movement of a mouse from the lower left to the upper right of a mouse pad.

The feature which is extracted from the signal 288, for example, may be features which designate some characteristic or feature which differentiates the curve representing the signal from other things. Various features are used in feature extraction methods, which are well known. In the present invention an exemplary feature is the smoothed power spectrum of the signal portion 288d. Other features may be the logarithm of the power spectrum, the power spectrum itself, etc.

The neural network 282 may be a conventional neural network trained in a typical fashion. The neural network 282 can compare the feature of the signal portion 288d with its training and can determine whether the feature is more like one trained feature or another trained feature. If the input feature of signal portion 288d is sufficiently like one trained feature, then the neural network causes production of an appropriate signal at the output, as was described above, for example. If the input feature is not sufficiently like any of the trained features or is relatively similar to two or more trained features, then the neural network will not cause the output to produce an output signal, for the uncertainty may otherwise cause production of an undesirable output signal.

As was described above, if two sensors 12 are used, then the system 270 may operate sequentially on each signal from respective sensors 12 or may include parallel processing capability so that the signals from both sensors can be processed substantially simultaneously. The output 284 may reflect the results of signal analysis of the signals originating from both sensors and may be used to provide an output signal at the output 284 that reflects in effect three dimensional data.

Another embodiment of the present invention is directed to a communication system, including a communications device and an ear microphone, or earset, assembly.

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The communications system allows a user to speak with a remotely located person. The ear microphone assembly, also referred to herein as an earset, includes a microphone and a speaker supported by a housing. The housing is retained by the ear of the user and allows for hands free use of the communication system while carrying on a conversation with the remotely located person. The microphone is directed towards the ear canal of the user and detects sounds emanating or coming out of the ear (or air pressure changes occurring within the ear) to accurately and reliably detect the speech of the user. The position of the microphone may also help to reduce detection of externally generated sounds. The ear microphone assembly allows for separation of a speech input device for a cellular, or wireless, telephone from an RF transceiver of the telephone.

Additionally, the earset assembly of the present invention allows a user to speak more quietly (e.g., such as at a whisper or near whisper) than with conventional headsets. This allows for more private conversations and less disruption to others. There is also a body of evidence indicating that the softer one speaks, the less concentration is needed to maintain the conversation, thereby allowing the individual at least to partially engage in other activities while speaking.

The earset of the present invention does not rely on directionality of the microphone or a detection of sound that has emanated from the user's mouth. Therefore, the need to repeatedly adjust the position of the earset that would otherwise distract the user and require the use of the user's hands. Also, the size and arrangement of the earset is small, resulting in a more cosmetically appealing device. Such a device can be used unobtrusively. For example, the device would not be noticed as much by others when used in public, or by a person being observed by others, such as, for example, a television news anchor or a secret service agent.

As a result, the sensor can be used to detect a person's speech. It is also noted that the term speech is used in its broadest sense and includes spoken words and utterances, as well as other vocalizations produced by the user, including, for example, grunts, whistles, singing, coughs, "clicking" sounds made by movement of the lips or tongue, and the like. To facilitate the description herein, the event of sensing or detecting by the microphone will be referred to as detecting and that which is detected will be referred to as a change within the ear, or simply an air pressure change. The present invention monitors changes within

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the human ear which occur instantaneously, or nearly instantaneously, in response to the speech of the person to provide a substantially real-time speech detection system.

Turning now to the figures, FIG. 15 is a block diagram which illustrates a communications system 32' for establishing duplexed (two-way) audio communication between two or more individuals. The communications system 32' includes a communications network 37, or backbone network. The communications network 37 establishes communication links with at least one communication device 36', thereby enabling a user of the communication device 36' to carry on a conversation with a remotely located person using a remotely located telephone 31 as is known in the art. In the illustrated embodiment, the communication device 36' and the remote telephone 31 are wireless telephones, such as cellular telephones. However, one skilled in the art will appreciate that the communication device 36' and/or the remote telephone 31 can be other types of devices, including hardwired (land line) telephones, radios, personal digital assistants (PDAs), portable or stationary computers, etc. The communications network 37 can also be alternative type of network, such as the Internet, a WAN, or a LAN.

The communications system 32' includes an ear microphone assembly, or earset 12', having a microphone 34' and a speaker 38' supported by a housing 150' (FIG. 16). The physical arrangement of the earset 12' will be described in more detail below. The microphone 34' is used to detect sounds emanating or coming from the ear (sometimes referred to as air pressure changes within the ear) of the user that result from speech of the user. The microphone 34' converts those detections into an electrical signal. The electrical signal is optionally amplified by an amplifier 24'. Next, the amplified electrical signal is received by the communication device 36' via an appropriate interface adapter 26'. The interface adapter 26' can be, for example, a jack or other electrical connector received by a corresponding receptacle of the communication device 36'.

The interior portion 154' of the housing 150' has a recess 156' in which the microphone 34' is placed. Preferably, the air adjacent the ear drum 112 and/or the air in the ear canal 110 will be in fluid communication with the air adjacent the microphone 34'. In this way, sounds or air pressure changes generated within the ear 100 will be transmitted directly to the microphone 34' through an air medium. Other sounds conducted by the bones and tissue of the user may be transmitted to the microphone 34' mechanically via the housing 150'. In one embodiment, the microphone 34' is held in place with an adhesive.

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The microphone 34' in one embodiment is less than two mm in size. In an alternative embodiment, the microphone 34' is disposed within a hollow cavity defined by the housing 150' and an opening in a wall of the housing 150' defining the cavity is provided to establish fluid communication of the air inside the cavity and the air inside the ear canal 110.

The interior portion 154' of the housing 150' has a second recess 157 in which the speaker 38' is placed. Preferably, the speaker 38' broadcasts sounds corresponding to the speech of the remotely located person using the remote telephone 31 and received by the communication device 36'. The sounds are a reproduction of the remotely located person's speech and are made audible to the user by the speaker 38'.

The microphone 34' is electrically coupled to the amplifier 24' with a conductor 158'. The speaker 38' is also electrically coupled to the amplifier 24' with a conductor 160'. The amplifier 24' is electrically coupled to the interface adapter 26' (FIG. 15) with a conductor 162'. More specifically, the conductor 162' carries the amplified signals from the microphone 34' to the interface adapter 26' and carries signals to be converted into sound from the interface adapter 26' to the speaker. As one skilled in the art will appreciate, the amplifier 24' can be omitted or by-passed and the microphone 34' and/or speaker 38' can be coupled directly to the interface adapter 26' by way of a conductor or conductors.

The speaker 38' is used to transmit (i.e., broadcast) sounds to the user. These sounds include sounds generated in response to signals received by the communication device 36' over the communications network 37. In this way, the earset 12' and communication device 36' can be used as a bi-directional communication apparatus.

In one embodiment, the earset 12' is coupled to the communication device using three conductors. A first of the conductors is used as a common ground. A second of the conductors carries electrical signals to the speaker 38' to be converted into sound waves. The third of the conductors carries electrical signals representing detected sounds from the microphone 34' to the communication device 36' for transmission to the communications network 37. The third conductor also carries power for the microphone 34' and amplifier 24'. In this embodiment, the interface adapter 26' can be implemented with a 2.5 mm jack as is known in the art.

It has been shown that the use of one earset 12' with the microphone 34' in one of the ears 100 of an individual can be used to detect speech such that the resulting

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transmission by the communication device 36' allows the remotely located person using the remote telephone 31 to hear the speech of the user. However, empirical testing has shown that by using two such earsets, one in each of the ears of an individual, the performance of system 32' is improved due to the detection of more information.

A block diagram of an ear microphone assembly, and particularly the amplifier 24', is illustrated in FIG. 17. Since the microphone 34' is placed in close proximity to the speaker 38', the microphone 34' will tend to detect sounds generated by the speaker 38'. These detections are in addition to detections of the user's speech. The sounds that have been generated by the speaker 38' and detected by the microphone 34' will be referred to herein as feedback and is represented in FIG. 17 by arrow 51. In one embodiment, the sensitivity of the earset 12' is tuned to minimize detection of sounds from the speaker 38' and/or to minimize the delivery of a feedback component to the communication device 36'. For example, a potentiometer can be used to adjust the gain of the amplifier 24' as desired. Physical separation of the speaker 38' and the microphone 34' to reduce the detection of feedback is also possible.

In the illustrated embodiment, the amplifier 24' is configured to cancel at least some, if not all, of the feedback component of the electrical signal generated by the microphone 34'. In this embodiment, the speaker 38' and the microphone 34' are placed in close physical proximity so that the feedback detected by the microphone 34' closely resembles the sound produced by the speaker 38'. FIG. 17 illustrates an exemplary amplifier circuit 24' for canceling at least a portion of the feedback component. As one skilled in the art will appreciate, a variety of alternative analog or digital circuits can be used, each falling within the scope of the present invention.

The illustrated amplifier circuit 24' receives a electrical signal output from the microphone 34', referred to herein a microphone signal. The microphone signal is filtered by a filter 53. The filter 53 can be, for example, a filter to remove low frequency components of the microphone signal (e.g., frequencies below about 400 Hz). The filtered microphone signal is then amplified by a amplifier 55, such as a low power, non-inverting operational amplifier. The gain of the amplifier 55 is selected to condition the microphone signal to a voltage level meeting the input requirements of the communication device 36'.

As indicated, the speaker 38' is coupled to the communication device 36' to receive an electrical signal for conversion into sound to be broadcast to the user of the earset 12'.

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This signal will be referred to herein as a speaker signal. The speaker signal is also filtered by a filter 58. The filter 58 can be, for example, a filter to remove low frequency components of the speaker signal (e.g., frequencies below about 400 Hz). The filtered speaker signal is then amplified by an amplifier 60. The amplifier 60 can be, for example, a low power, inverting operational amplifier. The gain of the amplifier 60 is selected to condition the speaker signal to have an amplitude equal to, or approximately equal to, the amplified feedback component of the microphone signal.

The amplified microphone signal and the amplified speaker signal are then combined so as to cancel the feedback component of the microphone signal and generate a corrected microphone signal. It is noted that the polarity or phase of the amplified microphone signal and/or the amplified speaker signal may be adjusted to correctly combine the signals. The combination of the signals can be carried out by an adder 56. As one skilled in the art will appreciate, in an analog system, the signals can be combined by simply wiring the signals together. In one example, the conductors carrying the signals are directly connected together. Alternatively, analog or digital circuitry or processing can be used to carry out the combining function of the adder 56.

It is noted that the speaker signal can be delayed by a delay element if needed to compensate for any propagation delay of the feedback 51. An output of the adder 56 is coupled to a filter 62 for further conditioning of the corrected microphone signal, such as removing high frequency components of the corrected microphone signal (e.g. frequencies above about 4 KHz for the communication system described herein).

Many alternatives to the illustrated amplifier circuit 24' are contemplated and each are considered to fall within the scope of the present invention.

As another example, the high and low frequency filters can be replaced by a band pass filter. In another example modification, the speaker signal can be amplified and/or filtered before being applied to the speaker.

In yet another example modification, the analog microphone and speaker signals can be converted to digital signals and processed using digital signal processing techniques to derive a microphone signal with no, or very little, feedback component.

Processing of the microphone signal by the communication device 36' for transmission is well known in the art and will not be described in greater detail.

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With additional reference to FIG. 18, in another embodiment of the present invention, the earset 12' can include an external microphone 54' in addition to the microphone 34' directed toward the ear canal of the user (also referred to herein as an internal microphone 34'). The external microphone 54' detects air pressure changes, or sounds, originating outside the user's ear. As described below in more detail, these sounds are converted into electrical signals and compared with electrical signals corresponding to the sounds detected by the internal microphone 34'. The respective signals can be processed to discern air pressure changes within the ear from externally produced noise, speech and the like.

The external microphone 54' is disposed on a exterior portion 72' of the housing 150' and is pointed away from the wearer of the earset 12' assembly. The external microphone 54' can be secured to the housing 150' with adhesive and can be disposed in a recess defined by the exterior portion 72'. Similar to the internal microphone 34', the external microphone 54' is small (e.g., less than two mm in size).

The position of the external microphone 54' allows the external microphone 54' to detect air pressure changes present near the wearer's ear but having a source external to the user of the earset 12'. In one embodiment, the external microphone 54' is an omnidirectional microphone.

FIG. 18 is a block diagram that illustrates a system 32" for detecting speech of the user. The system 32" includes the earset 12' having the internal microphone 34', the speaker 38' and amplifier 24a' as described in greater detail above. In addition, the earset 12' of the system 32" has the external microphone 54'. As one skilled in the art will appreciate, the use of the external microphone 54' is optional. The external microphone 54' is disposed on the housing 150' (FIG. 16) and generally points away from the person wearing the earset 12'. As a result, the external microphone 54' detects air pressure changes located outside the ear. These air pressure changes are mainly the result of externally generated noise (other people's speech, environmental noise, etc.) and, in part, to the speech of the person wearing the earset 12', when the speech travels from the mouth of the user and through the air to the external microphone 54'.

The external microphone 54' generates an electrical signal corresponding to the sounds detected by the external microphone 54'. The external microphone 54' is electrically coupled to the communication device 36' using a separate and distinct input, or channel,

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than is used for the internal microphone 34'. The signal generated by the external microphone can be amplified by an amplifier 24b' and coupled to the communication device 36' with an interface adapter 26b'. The communication device 36' correlates the two signals produced by the microphones 34', 54' as described below. Alternatively, the internal microphone 34' and the external microphone 54' are respectively connected to the left and right channel inputs of a processing circuit that correlates the signals as discussed below.

The unique signals generated by the internal and external microphone 34', 54' can be used to distinguish between user produced speech and externally produced sounds. In alternative configurations of the system 32", the system 32" can be adapted to cancel externally generated noise from the detections of the internal microphone 34'.

When the user speaks both the internal microphone 34' and the external microphone 54' detects the speech. Sounds generated by a source other than the user (e.g., another person speaking, a passing vehicle, machinery, the hum of an air conditioner or computer, etc.) will be referred to herein as noise. Both the external microphone 54' and the internal microphone 34' detect the noise. Accordingly, the communication device 36' (or other processing circuit) can be programmed to isolate the user's speech by identifying (i.e., separating) the speech component and the noise component. As a result, a relatively noise-free speech signal can be produced which can be transmitted to the communication network 37 (FIG. 15) by the communication device 36'. One skilled in the art will appreciate that the earset 12' can be used with other systems (e.g., the systems 10, 22, 32, 40 and 52 described above) to assist in separating an initiating action component and a noise component in the manner described herein for the system 32" (e.g., by using the method 300 described below).

In one embodiment, signal separation can be made by solving equations 1 and 2, identified below, for a sound component signal ("S") and a noise component signal ("N"). An impulse response of the internal microphone (F_{MI}) and an impulse response of the external microphone (F_{ME}) reflect the sensitivity of the microphones, the frequency response of the microphones, resonating characteristics of the ear and housing, etc. Y_I represents the overall signal produced by the internal microphone and Y_E represents the overall signal produced by the external microphone. Assuming that the entire system is

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linear (the system features are additive and homogeneous), the system can be represented by equations 3 and 4 as follows (the asterisk, or "*", represents a convolution operator):

$$F_{MI} * S + F_{MI} * N = Y_I \quad \text{Eq. 3}$$

$$F_{ME} * S + F_{ME} * N = Y_E \quad \text{Eq. 4}$$

By using the output signals Y_I and Y_E and the known impulse responses F_{MI} and F_{ME}, the two equations can be solved in real time for the two unknown signals S and N.

A method 300 for detecting speech using the system 32" is illustrated in FIG. 19. The method 300 includes the detection of changes in air pressure with both the internal microphone 34' and the external microphone 54'. More specifically, in step 302, changes in air pressure within the ear 100 are detected by the internal microphone 34'. These detections can be the result of the user speaking, but can also include a noise component generated external to the person wearing the earset 12' and/or a sound component generated by speech of the user and traveling from the user's mouth, through the air and to the user's ear 100 area. Also in step 302, changes in air pressure near, but outside, the ear 100 of the user are detected by the external microphone 54'. These detections are mainly the result of noise generated external to the user and/or a sound component generated by speech of the user and traveling from the user's mouth, through the air and to the user's ear 100 area. The detections are converted by the microphones 34' and 54' into respective electrical signals, herein referred to as an internal signal and an external signal. The internal signal can be processed to reduce any detected feedback from the speaker 38' as discussed in more detail above.

Next, in step 304, the speech and noise components are separated as described above. The resulting speech signal from step 304 is used in step 306 to provide the signal transmitted by the communication device 36'.

The diagrams described herein show the architecture, functionality and operation of various systems for detecting changes within a human ear and producing an output signal or control instruction corresponding to an initiating action. Each functional block may represent a module, segment, or portion of software code that contains one or more executable instructions to implement the specified logical function(s).

Although the block diagrams and flowcharts show a specific order of execution, it is understood that the order of execution may differ from that depicted. For example, the order of execution of two or more blocks may be altered relative to the order shown. Also,

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two or more blocks shown in succession may be executed concurrently or with partial concurrence. In addition, various blocks may be omitted. It is understood that all such variations are within the scope of the present invention.

Although the invention has been shown and described with respect to a certain preferred embodiment or embodiments, it is obvious that equivalent alterations and modifications will occur to others skilled in the art upon the reading and understanding of this specification and the annexed drawings. In particular regard to the various functions performed by the above described components (assemblies, devices, circuits, etc.), the terms (including a reference to a "means") used to describe such components are intended to correspond, unless otherwise indicated, to any component which performs the specified function of the described component (*i.e.*, that is functionally equivalent), even though not structurally equivalent to the disclosed structure which performs the function in the herein illustrated exemplary embodiments of the invention. In addition, while a particular feature of the invention may have been disclosed with respect to only one of several embodiments, such feature may be combined with one or more other features of the other embodiments as may be desired and advantageous for any given or particular application.

The foregoing embodiments are illustrative only of the principles of the invention, and various modifications and changes will readily occur to those skilled in the art. The invention is capable of being practiced and carried out in various ways and in other embodiments. It is also to be understood that the terminology employed herein is for the purpose of description and should not be regarded as limiting.

The term "comprise" and variants of the term such as "comprises" or "comprising" are used herein to denote the inclusion of a stated integer or stated integers but not to exclude any other integer or any other integers, unless in the context or usage an exclusive interpretation of the term is required.

Any reference to publications cited in this specification is not an admission that the disclosures constitute common general knowledge in Australia.

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THE CLAIMS DEFINING THE INVENTION ARE AS FOLLOWS:

1. An earset comprising:

a housing positionable with respect to an ear of a person;

a microphone disposed with respect to the housing for insertion into the ear of a person, the microphone operable to detect a change in air pressure within the ear while the person speaks and to produce an electrical microphone signal corresponding to the internally detected change in air pressure;

a speaker disposed with respect to the housing and operable to produce a sound corresponding to an electrical speaker signal input to the speaker; and

a circuit coupled to receive the microphone signal produced by the microphone and the electrical speaker signal as input to the speaker, and operable to produce a corrected microphone signal having a reduced feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound produced by the speaker wherein the circuit includes:

an amplifier for amplifying the microphone signal produced by the microphone;

an amplifier for amplifying the electrical speaker signal received by the circuit and outputting an electrical feedback indication signal; and

a combining means for combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

2. The earset according to claim 1, wherein the combining means is a signal adder.

3. The earset according to claim 1, wherein the microphone is disposed for placement in a concha of the ear.

4. The earset according to claim 1, wherein the speaker is disposed for placement in a concha of the ear.

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5. The earset according to claim 1, wherein the microphone is disposed for placement in an ear canal of the ear.

6. The earset according to claim 1, wherein the speaker is disposed for placement in an ear canal of the ear.

7. In combination, a communications system including the earset according to claim 1, and further comprising a communication device coupled to receive the corrected microphone signal and transmit an output corresponding to the corrected microphone signal to a remote location, and the communication device adapted to receive a signal from which the speaker signal is produced by the communication device.

8. The earset according to claim 1, further comprising a second microphone disposed with respect to the housing and located adjacent the ear and operable to detect a change in air pressure external to the ear of the person and to produce an electrical signal corresponding to the externally detected change in air pressure.

9. The earset according to claim 8, wherein a processing circuit executes logic to compare a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the electrical signal corresponding to the externally detected change in air pressure to distinguish detections of air pressure changes produced by the person's speech from air pressure changes produced by a source external to the person.

10. A method of detecting speech of a person, comprising the steps of:

placing a microphone at least partially within an ear of the person;

detecting a change in air pressure within the ear of the person using the microphone, the change in air pressure corresponding to the speech of the person;

producing an electrical microphone signal corresponding to the internally detected change in air pressure;

placing a speaker with respect to the ear of the person;

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generating a sound with the speaker corresponding to an electrical speaker signal input to the speaker; and

generating a corrected microphone signal by reducing a feedback component of the microphone signal, the feedback component resulting from the detection by the microphone of the sound generated by the speaker and wherein reducing the feedback includes:

amplifying the microphone signal produced by the microphone;

amplifying the electrical speaker signal to generate an electrical feedback indication signal; and

combining the amplified microphone signal and the feedback indication signal to produce the corrected microphone signal.

11. The method according to claim 10, wherein the microphone is placed in a concha of the ear.

12. The method according to claim 10, wherein the speaker is placed in a concha of the ear.

13. The method according to claim 10, wherein the microphone is placed in an ear canal of the ear.

14. The method according to claim 10, wherein the speaker is placed in an ear canal of the ear.

15. The method according to claim 10, further comprising the steps of:

coupling the corrected microphone signal to a communication device;

transmitting an output corresponding to the corrected microphone signal by the communication device;

receiving a signal from which the speaker signal is produced by the communication device; and

outputting the speaker signal by the communication device.

16. The method according to claim 10, further comprising the steps of detecting an air pressure change adjacent and external to the ear of the person with a

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second microphone and producing an electrical signal corresponding to the externally detected change in air pressure.

17. The method according to claim 16, further comprising the step of comparing a signal strength difference between the electrical signal corresponding to the internally detected change in air pressure and the electrical signal corresponding to the externally detected change in air pressure to distinguish detections of air pressure changes produced by the speech from air pressure changes produced by a source external to the person.

Dated this second day of February 2006

Think-A-Move, Ltd.

By their Patent Attorneys

CULLEN & CO.

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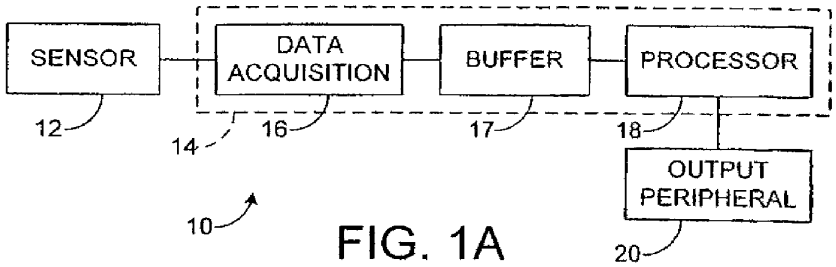


FIG. 1A

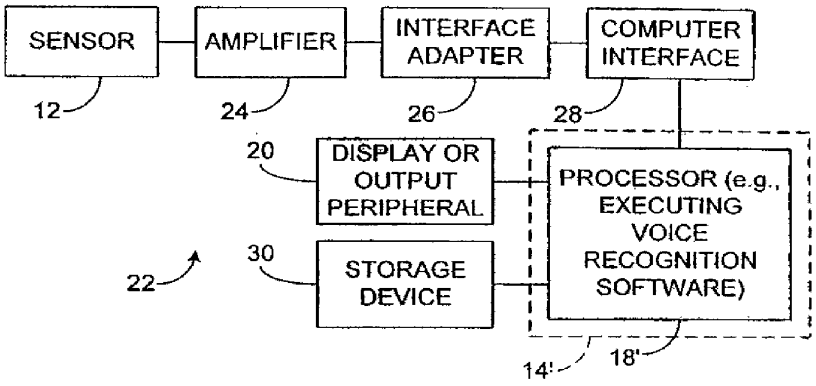


FIG. 1B

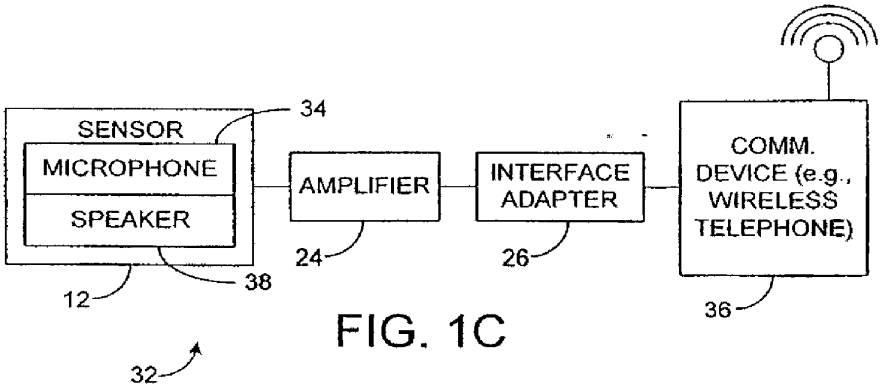


FIG. 1C

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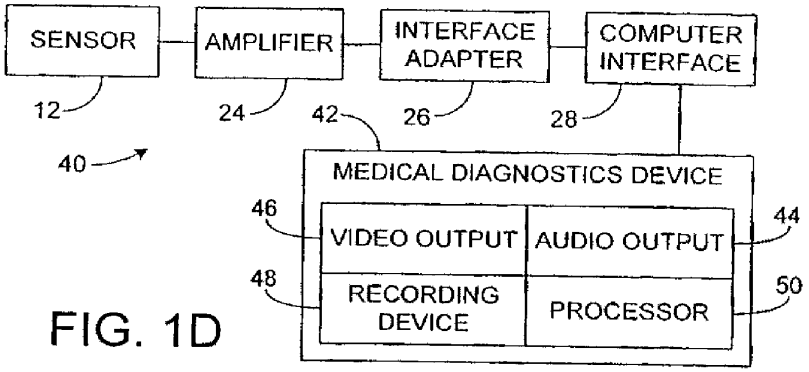


FIG. 1D

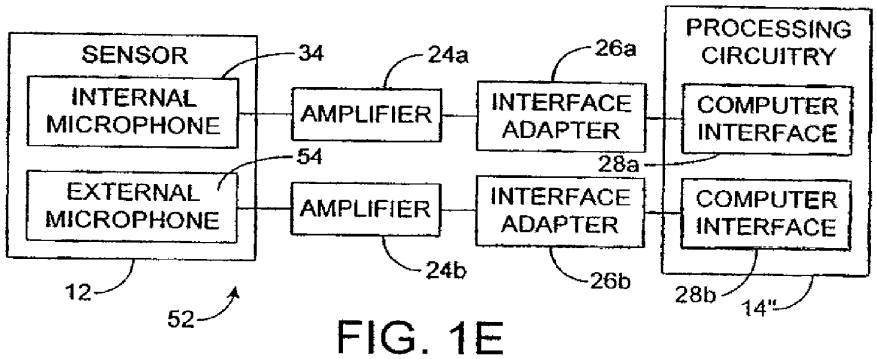


FIG. 1E

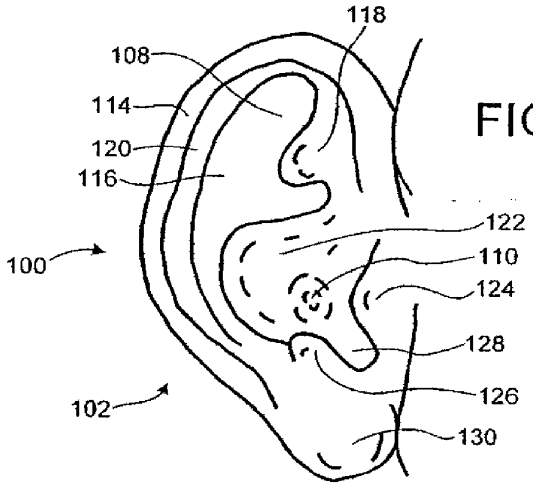


FIG. 2A

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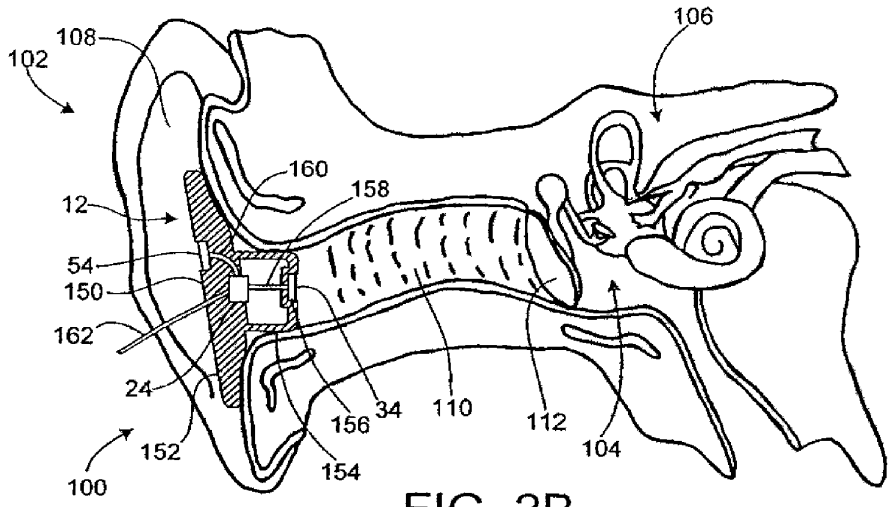


FIG. 2B

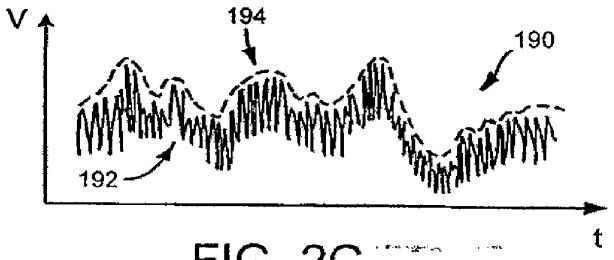


FIG. 2C

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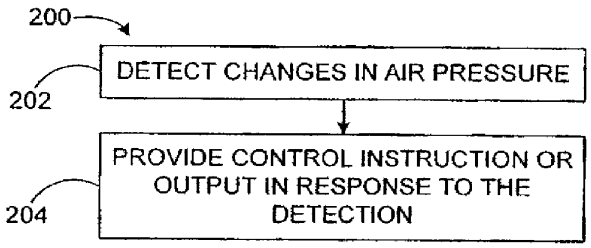


FIG. 3A

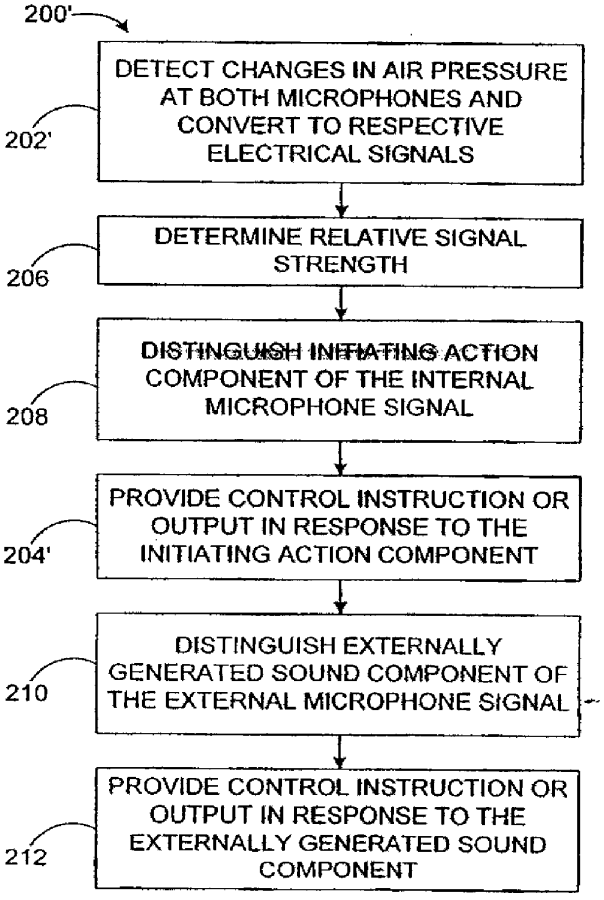


FIG. 3B

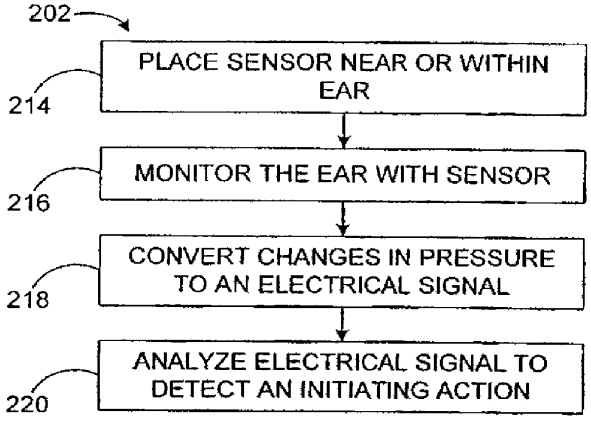


FIG. 4

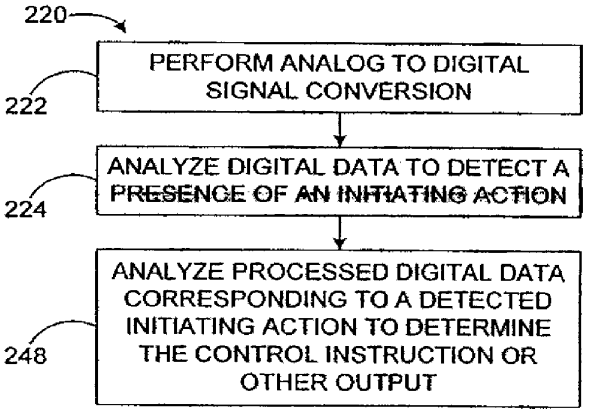


FIG. 5

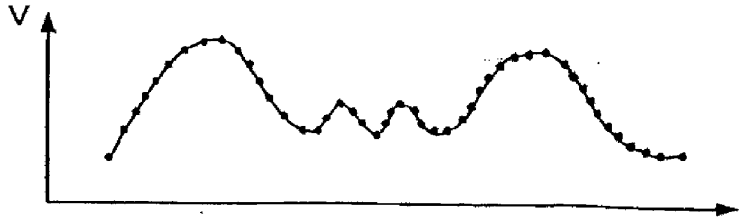


FIG. 6

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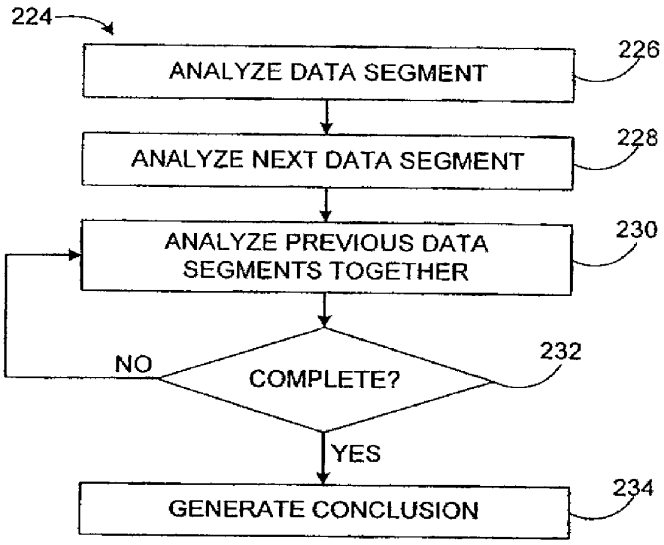


FIG. 7

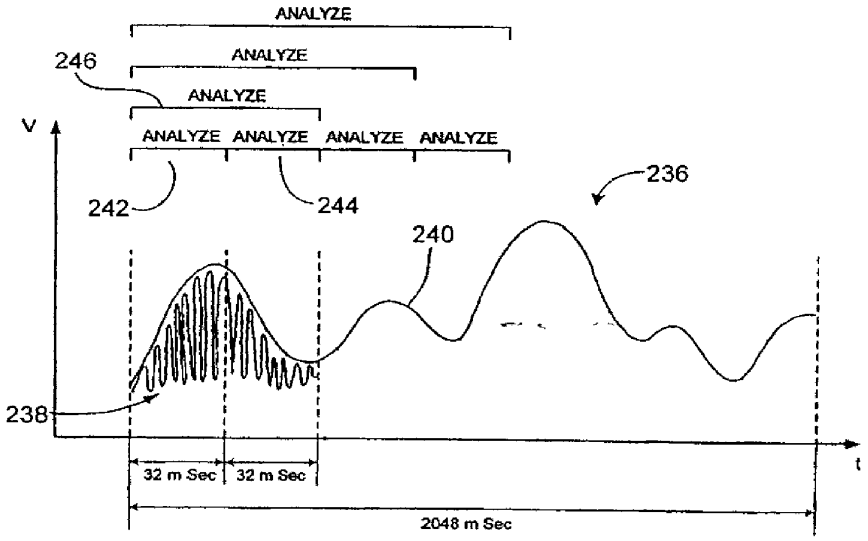


FIG. 8

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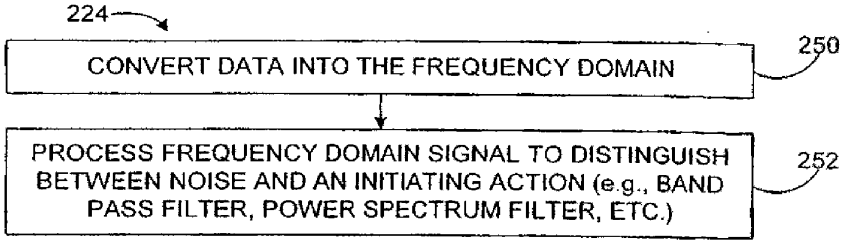


FIG. 9

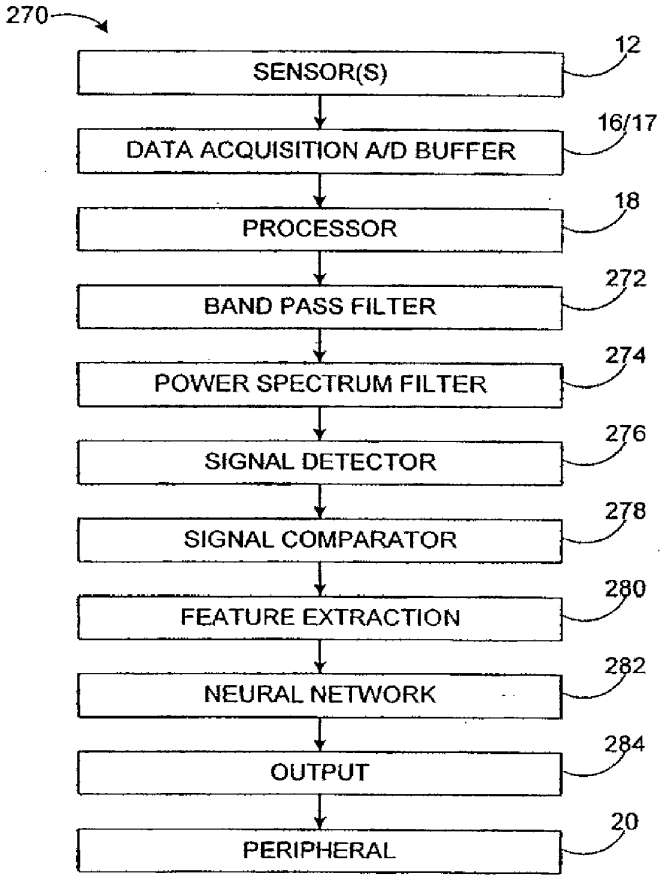


FIG. 10

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FIG. 11

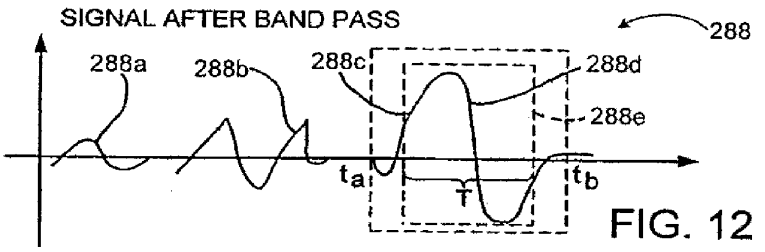


FIG. 12

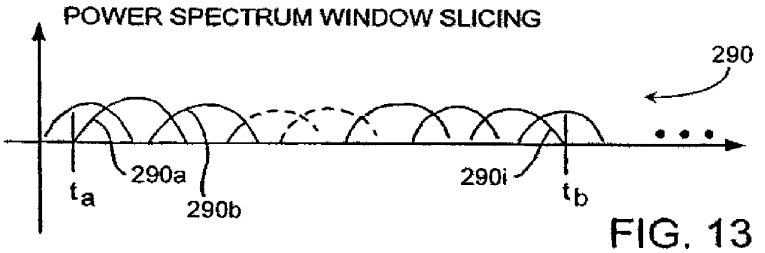


FIG. 13

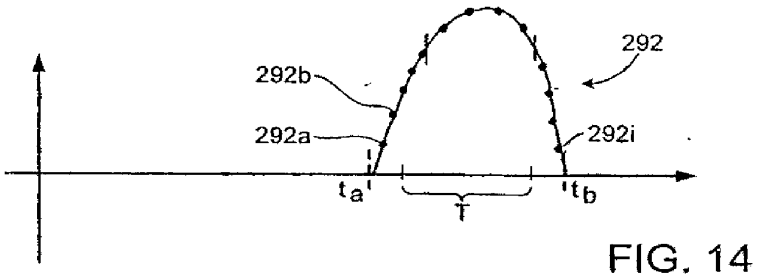


FIG. 14

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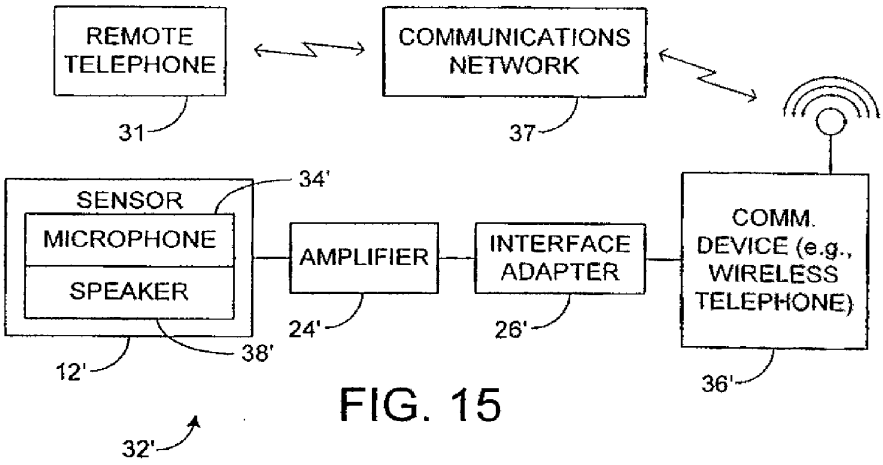


FIG. 15

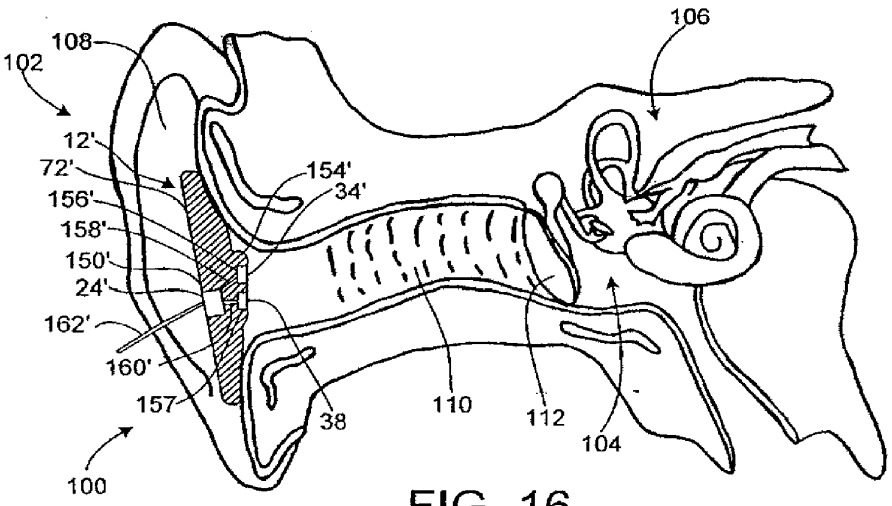


FIG. 16

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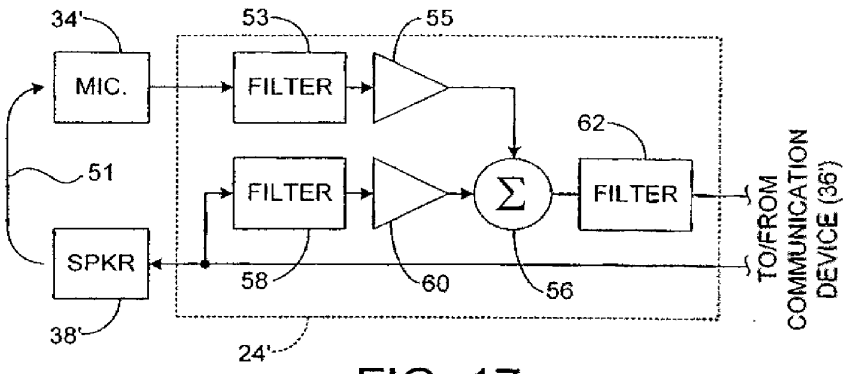


FIG. 17

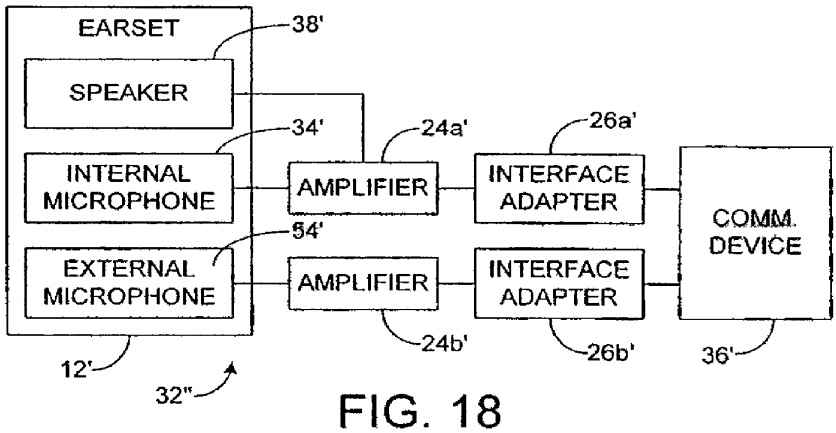


FIG. 18

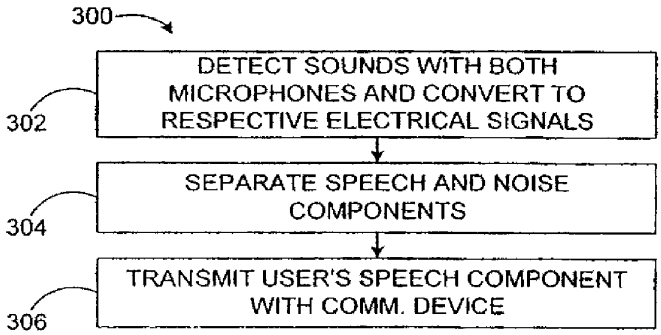


FIG..19

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European Patent Office
Office européen des brevets

(11) EP 1 681 903 A2

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(54) Wireless audio signal receiver device for a hearing instrument

(57) The invention relates to a receiver device (10) for receiving audio signals from a remote source (12), comprising: a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24) which is capable of being mechanically connected to an input interface (26) of a hearing instrument (16) to be worn at a user's ear in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein the antenna is designed as a printed board circuit with a loop-like conductor (56) on an at least partially flexible insolating substrate (55) and comprises a first portion (58) defining a first plane and a second portion (60) defining a second plane, wherein the first plane and the second plane are oriented at an angle of 60 to 120 degrees relative to each other.

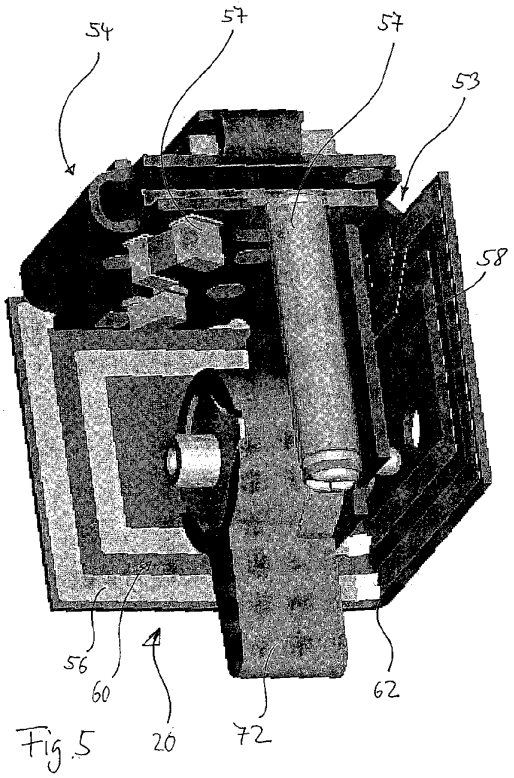


Fig. 5

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Description

[0001] The present invention relates to a wireless receiver device for wirelessly receiving audio signals from a remote source, which is capable of supplying such audio signal to a hearing instrument. The invention also relates to a hearing instrument which is capable of wirelessly receiving audio signals from a remote source.

[0002] It is well-known to use a receiver device (usually an FM (frequency modulation) receiver) for receiving audio signals from a remote source, for example a remote microphone, via a wireless link (usually an FM link) in order to provide such audio signals as input signals to a hearing instrument worn at a user's ear. To this end the receiver device has an output interface which is capable of being mechanically connected to an input interface of the hearing instrument via a so-called "audio shoe". The audio shoe is mechanically connected to the input interface of the hearing aid and comprises typically a standardized 3-pin socket for receiving three mating pins of the output interface of the receiver device, which pins typically are arranged in a line. However, the orientation of the three pins in the audio shoe with respect to the hearing instrument, i.e. with respect to the user's head, is not standardized and therefore varies from type to type. Moreover, the sensitivity of the antenna of the receiver device depends on the relative orientation to the user's head, with the optimum orientation of the antenna being given if the loop surface is perpendicular to the direction of the user's nose. If the receiver device is turned by 90°, the loss in sensitivity is typically of the order of 6 dB.

[0003] A presently used solution to this problem is to provide for a mechanical construction which allows to orient the receiver device in the optimum direction for all types of hearing instruments, wherein the connector, i.e. the mechanical components of the output interface of the receiver unit (i.e. the three pins), is rotatable with respect to the housing of the receiver device, so that prior to plugging the receiver unit into the audio shoe the connector can be rotated in an appropriate manner so as to adapt the orientation of the receiver device to the specific type of audio shoe / hearing instrument. Such receiver devices are sold, for example, by Phonak Communications AG, Murten, Switzerland, under the product designation MLx S.

[0004] A drawback of this solution is that a rotatable connector results in larger dimensions of the receiver device, given by the space required to design and implement a reliable and stable mechanical solution for the connector rotation. In addition, the electrical connections between the connector and the electronic module of the receiver device need to be flexible in order to allow the rotation of the connector with respect to the electronic module, which results in additional complexity, e.g. soldered wires, and again larger geometrical dimensions. A further drawback is the need to instruct the user regarding how to manipulate the receiver device, i.e. how to rotate the connector, on all sorts and types of combinations of hearing instruments and audio shoes.

[0005] EP 1 531 649 A2 describes a wireless hearing aid system with a magnetic loop antenna on a flex print, wherein at least a portion of the matching network is affixed to the flexible dielectric substrate carrying the antenna. The antenna may be attached to the inner or outer surface of the shell of the hearing aid, with the shape of the loop antenna being matched to the irregular shape of the hearing aid shell.

[0006] DE 10 2004 017 832 B3 relates to a hearing aid having a housing into which an antenna is integrated as an electrically conducting layer in order to reduce the size of the hearing aid. The antenna may be L-shaped as a metal layer applied to the hearing aid housing, the antenna may be applied as a pre-shaped foil element onto the hearing aid housing, the antenna may be produced by structuring a metal layer of the hearing aid housing or the antenna may be fabricated as a conducting plastics layer during injection moulding of the hearing aid housing.

[0007] EP 1 376 760 A2 relates to a folded dipole antenna for transmitting and receiving radio signals in all types of telecommunication systems, in particular for use in base stations of mobile telephone networks. The antenna consists of three portions, namely a central portion fixed to a ground plate, a left portion and a right portion which are angled by 45° to the central portion in such a manner that they form an angle of 90° relative to each other.

[0008] EP 1 594 187 A1 relates to a folded laminar antenna which is designed as a slot-loop antenna with a loop-like slot between conducting portions. The antenna consists of three portions, namely a central portion, a right portion and a left portion, with the right and left portions being folded by about 180° onto the central portion. The antenna comprises a layer of electrically conductive material which is provided on a dielectric substrate layer. The antenna may be used in portable wireless devices such as mobile telephones and personal digital assistants.

[0009] DE 10 2004 016 573 B3 relates to a binaural In-the-ear (ITE) hearing aid system wherein each of the two hearing aids has an antenna for wireless communication with the other hearing aid and wherein the antenna is oriented at a certain angle with respect to the housing in order to ensure that the two antennas are aligned when the hearing aids are worn by the user.

[0010] It is a first object of the invention to provide for a receiver device for being used with a hearing instrument, wherein the dependence of the sensitivity of the receiver device on the specific type of the hearing instrument should be relatively low while nevertheless the receiver device should be mechanically relatively simple.

[0011] It is a second object of the invention to provide for a receiver device for being used with a hearing instrument, wherein particularly simple manufacturing of the receiver device should be enabled.

[0012] It is a third object of the invention to provide for

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a hearing instrument capable of receiving audio signals from a remote source, wherein the space required by the antenna should be minimized.

[0013] According to the invention the first object is solved by a receiver device as defined in claims 1, 4 and 13, respectively, whereas the second object is achieved by a receiver device as defined in claim 22, and whereas the third object is achieved by a hearing instrument as defined in claims 11 and 12, respectively.

[0014] The solution according to claims 1 and 4 is beneficial in that, by using a magnetic loop antenna which comprises a first portion defining a first plane and a second portion defining a second plane, wherein the first plane and the second plane are oriented at an angle of 60 to 120°, it is ensured that at least one portion of the antenna is always oriented at least close to the optimum direction, whatever the actual orientation of the receiver device -depending on the type of hearing instrument to which it is to be connected - on the user's head will be. In addition, in view of the fact that usually the housing of such receiver device will have walls which are oriented relative to each other at an angle of typically between 60 and 120° the geometry of the magnetic loop antenna can be adapted to the geometry of the housing, whereby the volume of the receiver device can be reduced. In particular, also the need for a rotatable connector is avoided. Consequently, a more simple, smaller, more compact and also more robust receiver device can be achieved.

[0015] The solution according to claims 11 and 12 is beneficial in that, by providing a magnetic loop antenna which comprises a first portion defining a first plane and a second portion defining a second plane, wherein the first plane and the second plane are oriented at an angle of 60 to 120°, a relatively small, compact and robust hearing instrument can be achieved.

[0016] In a preferred embodiment of the solution according to claim 1 the first portion of the antenna is adjacent and aligned to a first wall of the housing and the second portion of the antenna is adjacent to and aligned to a second wall of the housing in order to achieve a particularly compact design. Preferably the first and second portion of the antenna are planar.

[0017] Preferably the first and second portion of the antenna unit are connected by a bent portion of the antenna unit, with the first and second portion of the antenna unit preferably comprising at least 80% of the area of the antenna. Preferably the first and second portion of the antenna have essentially the same area and are essentially symmetrical with respect to each other.

[0018] Preferably the angle between the first plane and the second plane is approximately 90°, for example from 80 to 100°.

[0019] In the case in which the antenna unit is designed as a printed board circuit with a loop-like conductor layer on a flexible insulating substrate, the device is preferably manufactured by forming the loop-line conductor layer on the flexible insulating substrate while the substrate is in a planar condition, bending the first and second portion relative to each other such that the first plane and the second plane become oriented at an angle of 60 to 120° and fixing the first and second portion in that orientation to each other, and mounting the antenna unit and the signal processing unit within the housing. Preferably the antenna unit and the signal processing unit are electrically connected prior to mounting the antenna unit and the signal processing unit within the housing, with the antenna unit and a preamplifier of the signal processing unit being tuned prior to mounting the antenna unit and the signal processing within the housing.

[0020] In the case in which the antenna unit is formed by an angled portion of the housing, the device is preferably manufactured by forming the housing with the integrated antenna unit and mounting the signal processing unit within the housing. Preferably the antenna unit is integrated into the housing by a moulded interconnect device (MID) technique.

[0021] According to one embodiment, the housing is shaped first and subsequently the antenna unit is integrated into the housing by modifying the surface of the housing. Preferably, the plastic material is capable of being made conductive by laser activation and the loop-like conductor is created by laser activation of a surface portion of the plastic material of the housing, followed by electroplating of the laser-activated surface portion in order to thicken the laser-activated surface portion.

[0022] According to an alternative embodiment the loop-like conductor is created by metal deposition from a metal evaporation source through a shadow mask onto a surface portion of the plastic material of the housing.

[0023] According to a further alternative embodiment the loop-like conductor is created by coating at least a portion of the surface of the housing with a metal layer, followed by selectively structuring the metal layer in order to create a loop-like metal structure, wherein the metal layer preferably is structured by selectively removing the metal layer.

[0024] According to another alternative embodiment the antenna is integrated into the housing during shaping of the housing, wherein the housing preferably is shaped by injection moulding in a moulding tool, wherein the loop-like conductor is inserted into the moulding tool and is overmoulded in the moulding tool.

[0025] The solution according to claim 13 is beneficial in that, by orienting the antenna plane or antenna direction, respectively, at an angle of 30 to 60 degrees, preferably at 40 to 50 degrees, with respect to the central symmetry plane of the output interface, it is ensured - due to the fact that the orientation of the input interface or the adapter interface (audio shoe) typically differs by ±90 degrees or ±180 degrees from type to type of the hearing instrument so that also the relative orientation of the receiver device when connected to the hearing instrument would differ by 90 degrees or 180 degrees depending on the type of hearing instrument - that the orientation of the antenna to the user's head anatomy, in particular to the direction of the user's nose, differs by

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significantly less than 90 degrees depending on the type of hearing instrument so that the worst case in which the antenna is oriented more or less parallel to the direction of the user's nose can be avoided.

[0026] Usually said output interface comprises three pins which are arranged in a line, with these pins defining the central symmetry plane. The antenna may be a magnetic loop antenna, a ferrite coil antenna or an air coil antenna.

[0027] The solution according to claim 22 is beneficial in that, by forming a printed board circuit antenna and at least a portion of the signal processing unit as an integral electronic unit on a common printed circuit board comprising an at least partially flexible insolating substrate which is capable of being partially folded for mounting the printed circuit board into the housing, manufacturing of the receiver device is made particularly simple, since the antenna and at least a portion of the signal processing unit can be processed as an integral electronic unit, while, due to the foldability of the substrate, nevertheless a compact design can be achieved.

[0028] In the following examples of the invention will be illustrated by reference to the attached drawings, wherein:

Fig. 1 is a block diagram of an embodiment of a receiver device according to the invention when connected with a hearing instrument;

Fig. 2 is a block diagram of an embodiment of a hearing instrument according to the invention which is capable of receiving audio signals from a remote source;

Fig. 3 shows a flexprint assembly comprising an antenna for use in a receiver device to be connected to a hearing instrument or in a hearing instrument, with the assembly being shown in its original unfolded state;

Fig. 4 is a perspective view of the flexprint assembly of Fig. 4 after having been folded for being mounted in the housing of the receiver device or the hearing instrument, respectively;

Fig. 5 is a view similar to that of Fig. 4, with another embodiment of a folded flexprint assembly being shown;

Fig. 6 is an exploded view of a receiver device comprising the folded flexprint assembly of Fig. 5, a housing and a plug member;

Fig. 7 is a side view of a hearing instrument with a receiver device being connected thereto via an audio shoe;

Fig. 8 shows an example of part of a housing of a receiver device or hearing instrument, with an antenna being integrated within the walls;

Fig. 9 is a schematic view of a receiver device comprising a non-angled antenna and of a receiver device comprising an angled antenna when used with four different types of audio shoes.

[0029] Fig. 1 shows a block diagram of a receiver device 10 capable of wirelessly receiving audio signals from a remote source 12, which is connected via an audio shoe 14 to a hearing instrument 16 which may be a behind-the-ear (BTE) hearing aid which is worn at the user's ear. The remote audio signal source typically may be a transmitter unit comprising a microphone which is worn by a teacher in a classroom for teaching hearing-impaired pupils.

[0030] The receiver device 10 comprises a housing 18, a magnetic loop-antenna 20 for receiving radio-frequency signals carrying audio-signals from the remote source 12 via a radio-frequency link 22, a signal processing unit 21 for reproducing audio signals from the radio frequency signals received by the antenna 20, and an output interface 24 which is capable of being mechanically connected to an input interface 26 of the hearing instrument 16 via the audio shoe 14 which comprises an input interface 25 mating with the output interface 24. The signal processing unit 21 comprises a high frequency (HF) unit 29 connected to the antenna 20, a demodulator 30 for demodulating the frequency-modulated (FM) signal received by the antenna 20 and processed by the HF-unit 29, and a pre-amplifier 32 for pre-amplifying the demodulated audio signal prior to being passed to the output interface 24. The HF-unit 29 usually comprises a matching network for the antenna 20, an low-noise amplifier, an RF-amplifier, a frequency synthesiser and a mixer in order to convert the HF-signal received by the antenna 20 down an intermediate frequency. The architecture of the receiver device described so far is conventional FM receiver architecture.

[0031] The hearing instrument 16 comprises at least one microphone 34, a signal processing unit 36, an output transducer 38 (e.g. a loudspeaker) for stimulating the user's hearing, and a housing 40 and a battery 42 which typically also serves to power the receiver device via the audio shoe 14. When being used with the receiver unit 10, the hearing instrument 16 usually will have two different modes of operation: a first mode in which only the input audio signal received from the receiver device 10 is reproduced by the output transducer 38 (usually labelled "FM" mode) and a second mode in which a combination of the signal of the microphone 34 and the input signal provided by the receiver device 10 is reproduced by the output transducer 38 (usually labelled "FM+M" mode).

[0032] Fig. 2 is a block diagram of a hearing instrument 50, for example, a BTE hearing aid, which is capable of receiving audio signals form a remote source 12 via an

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FM link 22. To this end, a magnetic loop antenna unit 20 and a first signal processing unit 21 comprising a HF-unit 29, a demodulator 30 and a preamplifier 32 are integrated within the housing 52 of the hearing instrument 50. The system of Fig. 2 is functionally equivalent to the system shown in Fig. 1 in that audio signals from the remote source 12 can be provided to the user wearing the hearing instrument via the output transducer 38, with the functional components of the receiver device 10 of Fig 1 being mechanically integrated within the hearing instrument 50.

[0033] In Fig. 3 an example of an antenna 20 to be used for the receiver device 10 of Fig. 1 or the hearing instrument 50 of Fig. 2 is shown. The antenna 20 is part of an assembly 54 comprising a printed circuit board 53 which comprises an insulating substrate 55 which is flexible such that it is capable of being bent by at least 90°. As an alternative, the insulating substrate 55 may comprise rigid portions which are connected by flexible portions, i.e. in this case the substrate is only partially flexible. The assembly 54 comprises a loop-like conductor 56 on the insulating substrate 55, which forms two turns in order to form a magnetic loop antenna (however, also one turn may be already sufficient or three turns may be needed to achieve the optimum impedance). The area surrounded by the conductor turns should be as large as possible. The antenna 20 needs to have low resistance, which may be achieved by metalizing all layers of the print (or at least the uppermost layer and the lowermost layer (for a single layer print)) and realizing a conductive connection therebetween through an appropriate number of metalized via holes.

[0034] In addition to the antenna 20 the assembly 54 includes other electronic components 57 (ICs and passive components) forming at least part, preferably all, of the signal processing unit. In particular, the assembly will include at least the components of the HF-unit 29. Some of such electronic components 57 may be formed directly on the substrate 55 as conductor layers while others may be mounted as separate components on the substrate 55. The electronic components 57 formed directly on the substrate 55 may be formed in the same processing step as the antenna 20, whereas the separate components will be mounted thereafter. Thus the assembly 54 serves as an integral electronic unit, i.e. as an electronic module.

[0035] In Fig. 3 the assembly 54 is shown in a planar condition after manufacturing.

[0036] The antenna 20 comprises a first portion 58 and a second portion 60 which are connected by an intermediate portion 62 which allows the first and second portion 58, 60 to be folded with respect to each other so that they form an angle of, e.g. 90°, with the intermediate portion 62 being bent. The intermediate portion 62 may be of the same material as the first and second portion 58, 60 (if these are made of sufficiently flexible material) or it may be of a more flexible material of the substrate 55.

[0037] In addition, the assembly 54 comprises other portions 59, 61 carrying electronic components 57, which are foldable be about 90 degrees or 180 degrees relative to a central portion 63 in order to minimize the space required by the assembly 54.

[0038] Such folded configuration is shown in Fig. 4. Usually the intermediate portion 62 will be relatively small, for example, less than 20% of the area of the antenna unit 20. Usually the first and second portion 58, 60 of the antenna will have essentially the same area and preferably will be essentially symmetrical with respect to each other. After the assembly has been brought - manually or automatically - into the folded condition shown in Fig. 4 it may be mounted within the housing 64 shown schematically in Fig. 4 (the housing 64 may correspond to the housing 18 of the receiver device 10 or the housing 52 of the hearing instrument 50). The assembly 54 will be mounted in the housing 64 in such a manner that the first portion 58 is adjacent to and aligned to a first wall 66 of the housing 64 and that the second portion 60 is adjacent to and aligned to a second wall 68 of the housing 64, whereby a particularly compact design resulting in a small volume can be achieved.

[0039] A further benefit consists in the fact that due to the angled configuration of the antenna 20 it is ensured that one half of the antenna is always directed into the optimum orientation with respect to the user's head (antenna plane usually perpendicular to the user's nose), whatever the orientation of the pins/socket of the audio shoe 14 relative to the housing 40 of the hearing instrument 16 may be. Preferably the antenna 20 is electrically connected to the respective signal processing unit 21 comprising the pre-amplifier 32 prior to mounting to the antenna 20 and the signal processing unit 21 within the housing 64, so that the antenna 20 with the HF-unit 29 and the pre-amplifier 32 can be trimmed prior to being mounted in the housing 64 (i.e. the resonant radio frequency circuits will be tuned in order to account for parasitic capacitances and inductances).

[0040] In Fig. 5 another example of a flexprint antenna assembly 54 is shown in its folded state prior to being mounted within the housing, which includes a switch 72 for manually switching between the operation modes FM and FM+M of the hearing instrument 16 connected to receiver device 10.

[0041] In Fig. 6 the folded flexprint assembly 54 of Fig. 5 (however, the conductor loops 56 of the antenna 20 have been omitted in Fig. 6) is shown together with a housing 64 and a plug member 74. The plug member 74 comprises three pins 76A, 76B, 76C which form the output interface 24 of the receiver unit 10 and which are arranged in a line so that they define a central symmetry plane of the output interface 24 (however, the pins 76A and 76C are of different diameters in order to ensure the plug member 74 can be connected to the audio shoe 14 only in the correct orientation—and not in an orientation rotated by 180 degrees with respect to the correct orientation). The plug member 74 also comprises a carrier unit 70 for receiving the folded assembly 54. For assembling the receiver device 10 the folded assembly 54 with the carrier unit is fixed at the carrier unit 70 of the plug mem-

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ber 74. The flexprint assembly 54 comprises contacts 75 which engage with the inner ends of the pins 76A, 76B, 76C upon when the assembly 54 is connected to the plug member 74. Finally the folded assembly 54 fixed at the plug member 74 is mounted within the housing 64. To this end the plug member 74 is fixed to the housing 64 by two rods 77, with the plug member 74 forming the cover plate of the housing 64.

[0042] In Fig. 7 an arrangement is shown in which a receiver unit 10 comprising a housing 18 is connected via an audio shoe 14 to a hearing instrument 16, with the switch 72 projecting through the housing 18 for being operated by the user.

[0043] It is to be understood that, while the first portion 58 and the second portion 60 of the antenna 20 are shown in Figs. 4 and 5 as being planar, this need not be necessarily so. In particular, if for design reasons the walls of the housing in which the antenna 20 is to be mounted have a rounded shape, the first portion 58 and the second portion 60 of the antenna 20 may have a correspondingly rounded shape. In this case the angle between the first and second portion of the antenna in the folded state may be determined by the angle between the respective tangential planes at the two opposite ends of the antenna (for example, if the antenna is bent in an "arch-like" manner in order to conform the shape of the antenna to a cylindrically shaped housing wherein the tangential planes at the two opposite ends of the bent antenna would form an angle of 90°, then the actual folding angle is 90°).

[0044] Fig. 8 shows an example of a housing 80 (only partially illustrated in the drawing) to be used for the receiver device 10 or for the hearing instrument 50 which capable of receiving audio signals from the remote source 12, into which housing 80 a magnetic loop antenna 20 has been integrated. As in the previous embodiments the antenna 20 comprises a loop-like conductor 56, with the conductor 56 being integrated into the walls of the housing 80. In the example shown in Fig. 7 the conductor 56 is integrated into a first wall portion 82 and a second wall portion 84, which are arranged at an angle of, for example, about 90° relative to each other. The antenna 20 is formed in such a manner that one half of the antenna is integrated into the first wall portion 82 while the second half of the antenna is integrated into the second wall portion 84, so that the two halves of the antenna, i.e. the first portion 58 of the antenna is oriented at an angle of about 90° relative to the second portion 60 of the antenna.

[0045] The conductor 56 may be formed, for example, by one of the following methods within the housing 80 which is made of plastic material:

[0046] The housing is shaped first (for example, by injection moulding) and subsequently the conductor 56 is formed by modifying the surface of the housing 80. One possibility to achieve this is to use a plastic material for the housing 82 which is capable of being made conductive by laser activation, wherein the conductor 56 is created by laser activation of the respective surface portion of the plastic material of the housing 80, followed by electroplating of the laser activated surface portion in order to thicken the laser activated surface portion. According to an alternative process, the conductor 56 may be created by metal deposition from a metal evaporation source through a shadow mask onto a surface portion of the plastic material of the housing 80. According to another alternative process, the conductor 56 is created by coating at last a portion of the surface of the housing 80 with a metal layer, followed by selectively structuring the metal layer into the desired shape of the conductor 56, preferably by selectively removing the metal layer.

[0047] Rather than shaping the housing first and subsequently integrating the antenna structure, the antenna structure, i.e. the conductor 56, may be integrated into the housing 80 during shaping of the housing. This can be done, for example, by shaping the housing by injection moulding in a moulding tool, wherein the conductor is inserted into the moulding tool and is overmoulded in the moulding tool.

[0048] All of these techniques are known as moulded interconnect device (MID) techniques.

[0049] Fig. 9 is a schematic view of a receiver device 10 comprising an angled antenna 20 and of a receiver device 110 comprising a non-angled antenna 120, respectively, when used with four different types of audio shoes. In Fig. 9 the respective orientation of the antenna 20, 120 with respect to the direction 94 of the user's nose 96 is shown, with the direction 94 extending between the ears 98 through the nose tip.

[0050] The receiver device 10 comprises an essentially rectangular housing 18 with a plug member comprising three pins 76A, 76B and 76C which are arranged in a line, thereby defining a central symmetry plane 90 of the output interface of the receiver device 10. The angled antenna 20 is of the type shown in Figs. 4 and 5, i.e. it comprises a first portion 58 which is angled by 90° relative to a second portion 60. In a first orientation labelled "A" in Fig. 9 the receiver unit 10 is used with a hearing instrument having an audio shoe of a first type which is oriented such that, when the receiver device 10 has been connected to the audio shoe and the hearing instrument is worn at the user's ear 98, the central plane 90 of the output interface of the receiver device 10 is perpendicular to the direction 94 of the user's nose 96. In this configuration, the first portion 58 of the antenna 20 likewise is oriented perpendicular to the direction 94 of the user's nose 96 so that the first portion 58 has an optimum orientation with respect to the user's head anatomy, while the second portion 60 has the least preferred orientation. In total, the antenna 20 thus will have medium sensitivity.

[0051] In the configuration labelled "B" in Fig. 9 the receiver device 10 is used with a different type of hearing instrument/audio shoe so that, when the receiver device 10 has been connected to the audio shoe and the hearing instrument is worn at the user's ear 98 the receiver device 10 has been rotated by 90° in the counter-clockwise di-

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rection compared to configuration A, so that the central symmetry plane 90 of the output interface now is parallel to the direction 94 of the user's nose 96. In this case the second portion 60 of the antenna 20 has the optimum orientation with respect to the direction 94 of the user's nose 96 while the first portion 58 now has the least preferred orientation. In total, however, the antenna performance thus is the same as in configuration A.

[0052] In the configuration labelled "C" the type of hearing instrument/audio shoe is such that the receiver device 10 has been rotated by 90° in the counter-clockwise direction compared to configuration B so that the central symmetry plane 90 now has the same orientation as in configuration A. Due to the 90° bent shape of the antenna 20, the performance of the antenna 20 is the same as in configurations A and B.

[0053] In the configuration labelled "D" the type of hearing instrument/audio shoe is such that the receiver device 10 has been rotated by 90° in the counter-clockwise direction compared to configuration C so that the central symmetry plane 90 now has the same orientation as in configuration B. Due to the 90° bent shape of the antenna 20, the performance of the antenna 20 is the same as in configurations A, B and C.

[0054] Consequently, by using an angled antenna 90, the performance of the antenna 20 is substantially independent of the specific type of hearing instrument/audio shoe with which the receiver device 10 is used.

[0055] An alternative embodiment in order to achieve such independence of antenna performance from the type of audio shoe is to use an antenna 120 which is either planar, thereby defining an antenna plane 92, or has an axial symmetry, thereby defining an antenna direction 92, wherein the antenna plane 92 or the antenna direction 92, respectively, is oriented at an angle of 30 to 60°, preferably from 40 to 50°, with respect to the central symmetry plane 90 of the output interface. If the antenna 120 is planar, it is preferably a magnetic loop antenna, whereas if it has an axial symmetry, it is preferably a ferrite antenna or an air coil antenna. Most preferably, the angle between the antenna direction 92 and the symmetry plane 90 of the output interface is about 45° as shown in Fig. 9. In this case, in configuration A, i.e. with the pins 76A to 76C being oriented such that the central symmetry plane 90 defined thereby is perpendicular to the direction 94 of the user's nose 96, the angle between the antenna direction 92 and the central symmetry plane 94 is 45°, resulting in medium performance of the antenna 120 compared to an orientation in which the antenna plane 92 or the antenna direction 92 would be perpendicular to the direction 94 of the user's nose 96.

[0056] In configuration B in which the orientation of the central symmetry plane 90 of the output interface has changed by 90° with respect to the direction 94 of the user's nose 96 due to the different type of audio shoe, the antenna direction 92 likewise has been rotated in the counter-clockwise direction by 90°. However, due to the angle of 45° between the antenna direction 92 and the central symmetry plane 90 of the output interface, the angle between the direction 94 of the user's nose 96 and the antenna direction 92 still is 45°. Consequently, the antenna performance will remain the same as in configuration A

[0057] This also applies to configurations C and D in which the antenna 120, due to the 45° orientation with respect to the central symmetry plane 90, has the same orientation with respect to the user's head 91 as in configurations A and B, respectively.

[0058] Thus, by using an antenna 120 which is oriented such that the angle of the antenna direction 92 with respect to the central symmetry plane 90 of the output interface is around 45°, the antenna performance is essentially independent of the specific type of hearing instrument/audio shoe with which the receiver device 110 is used.

[0059] The housing 18 shown in Fig. 9, which corresponds to the housing 64 of Fig. 6, has a four-fold rotational symmetry with respect to an axial symmetry axis and comprises two walls which are parallel to the central symmetry plane 90 of the output interface and two walls which are perpendicular to the central symmetry plane 90.

[0060] Generally, apart from the different design of the antenna 120, the receiver device 110 may have the same architecture as the examples of the receiver device described so far.

[0061] Also shown in Fig. 9 is a schematic example of the input interface 25 of the audio shoe 14, which comprises three pin sockets 79A, 79B, 79C for receiving the pins 76A, 76B and 76C, respectively, which sockets are arranged in a line and thereby define a central symmetry plane 93 of the input interface 25. The input interface 25 shown in Fig. 9 is an example of an audio shoe of the type resulting in the configuration "A" of the receiver devices 10, 110 of Fig. 9.

Claims

1. A receiver device (10) for receiving audio signals from a remote source (12), comprising: a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24) which is capable of being mechanically connected to an input interface (26) of a hearing instrument (16) to be worn at a user's ear in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein the antenna is designed as a printed board circuit with a loop-like conductor (56) on an at least partially flexible insolating substrate (55) and comprises a first portion (58) defining a first plane and a second portion (60) defining a second

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plane, wherein the first plane and the second plane are oriented at an angle of 60 to 120 degrees relative to each other.

2. The receiver device of claim 1, wherein the first portion (58) of the antenna (20) is adjacent to and aligned to a first wall (66) of the housing (64) and the second portion (60) of the antenna is adjacent to and aligned to a second wall (68) of the housing.

3. The receiver device of one of claims 1 and 2, wherein the first (58) and second portion (60) of the antenna (20) are formed on a rigid portion of the substrate (55) and are connected by a bent portion (62) formed on a flexible portion of the substrate.

4. A receiver device (10) for receiving audio signals from a remote source (12), comprising: a housing (18, 80) made of plastic material, a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24) which is capable of being mechanically connected to an input interface (26) of a hearing instrument (16) to be worn at a user's ear in order to supply the audio signals from the signal processing unit as input to the hearing instrument, wherein the housing encloses the signal processing unit, wherein the antenna is formed by an angled portion (82, 84) of the housing into which a loop-like conductor (56) forming a first antenna portion (58) defining a first plane and a second antenna portion (60) defining a second plane is integrated, and wherein the first plane and the second plane are oriented at an angle of 60 to 120 degrees relative to each other.

5. The receiver device of claim 4, wherein the first (58) and second portion (60) of the antenna (20) are connected by a bent portion (62) of the antenna.

6. The receiver device of one of the preceding claims, wherein the first (58) and second portion (60) of the antenna (20) comprise at least 80% of the area of the antenna.

7. The receiver device of claim 6, wherein the first (58) and second portion (60) of the antenna (20) have essentially the same area.

8. The receiver device of claim 7, wherein the first (58) and second portion (60) of the antenna (20) are essentially symmetrical with respect to each other.

9. The receiver device of one of the preceding claims, wherein the conductor (56) comprises one to three turns.

10. The receiver device of one of the preceding claims, wherein the angle between the first plane and the second plane is from 80 to 100 degrees.

11. A hearing instrument (50) to be worn at a user's ear and capable of receiving audio signals from a remote source (12), comprising: a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output transducer (38) for stimulating the user's hearing according to the reproduced audio signals, and a housing (52, 64) enclosing the antenna and the signal processing unit, wherein the antenna is designed as a printed board circuit (54) with a loop-like conductor (56) on an at least partially flexible insolating substrate (55) and comprises a first portion (58) defining a first plane and a second portion (60) defining a second plane, wherein the first plane and the second plane are oriented at an angle of 60 to 120 degrees.

12. A hearing instrument (50) to be worn at a user's ear and capable of receiving audio signals from a remote source (12), comprising: a housing (52, 80) made of plastic material, a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output transducer (38) for stimulating the user's hearing according to the reproduced audio signals, wherein the housing encloses the signal processing unit, wherein the antenna is formed by an angled portion (82, 84) of the housing into which a loop-like conductor (56) forming a first antenna portion (58) defining a first plane and a second antenna portion (60) defining a second plane is integrated, and wherein the first plane and the second plane are oriented at an angle of 60 to 120 degrees.

13. A receiver device (10) for receiving audio signals from a remote source (12), comprising: an antenna (120) for receiving radio frequency signals carrying audio signals, which antenna is either planar, thereby defining an antenna plane (92), or has an axial symmetry, thereby defining an antenna direction (92), a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24, 74, 76A, 76B, 76C) which is capable of being mechanically connected to an input interface (26) of a hearing instrument (16) to be worn at a user's ear (98) in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein the output interface has a central symmetry plane (90), and wherein

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said antenna plane or antenna direction is oriented at an angle of 30 to 60 degrees with respect to said central symmetry plane of the output interface.

14. The receiver device of claim 13, wherein said angle is from 40 to 50 degrees.

15. The receiver device of one of claims 13 and 14, wherein said output interface (24, 74) comprises three pins (24, 74, 76A, 76B, 76C) which are arranged in a line, and wherein said pins define said central symmetry plane (90).

16. The receiver device of one of claims 13 to 15, wherein said antenna (120) is a magnetic loop antenna, a ferrite antenna or an air coil antenna.

17. The receiver device of one of claims 13 to 16, wherein said housing (64) has a four-fold rotational symmetry with respect to an axial symmetry axis and comprises two walls which are parallel to said central symmetry plane (92) of said output interface (74, 76A, 76B, 76C) and two walls which are perpendicular to said central symmetry plane of said output interface.

18. A system comprising a receiver device (10) for receiving audio signals from a remote source (12) and a hearing instrument (16) worn at a user's ear, said receiver device comprising: an antenna (120) for receiving radio frequency signals carrying audio signals, which antenna is either planar, thereby defining an antenna plane (92), or has an axial symmetry, thereby defining an antenna direction (92), a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24, 74, 76A, 76B, 76C) which is capable of being mechanically connected directly or via an interface (25) of an adapter (14) to an input interface (26) of said hearing instrument (16) in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein the output interface has a central symmetry plane (90), wherein said antenna plane or antenna direction is oriented at an angle of 30 to 60 degrees with respect to said central symmetry plane of the output interface, and wherein said input interface or said adapter interface has a central symmetry plane (93) which is oriented, when said hearing instrument is worn at the user's ear (98), at an angle of 30 to 60 degrees with respect to the direction (94) of the user's nose (96).

19. The system of claim 18, wherein said angles are from 40 to 50 degrees.

20. The system of one of claims 18 and 19, wherein said input interface (26) or said adapter interface (25) comprises three pin sockets (79A, 79B, 79C) which are arranged in a line, and wherein said pin sockets define said central symmetry plane (93).

21. The system of one of claims 18 to 20, wherein said hearing instrument (16) is a behind-the-ear (BTE) hearing instrument.

22. A receiver device (10) for receiving audio signals from a remote source (12), comprising: a magnetic loop antenna (20) for receiving radio frequency signals carrying audio signals, a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24) which is capable of being mechanically connected to an input interface (26) of a hearing instrument (16) to be worn at a user's ear in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein the antenna and at least a portion of the signal processing unit are formed as an integral electronic unit (54) on a common printed circuit board (53) comprising an at least partially flexible insolating substrate (55) which is capable of being partially folded for mounting the printed circuit board within the housing, wherein said portion of the signal processing unit comprises at least a high frequency (HF)-unit (29) receiving HF-signals from the antenna and providing signals at intermediate frequencies to a demodulator (30) of the signal processing unit and wherein the antenna is designed as a printed board circuit with a loop-like conductor (56) on the insolating substrate.

23. The receiver device of claim 22, wherein said portion of the signal processing unit (21) comprises in addition said demodulator (30).

24. The receiver device of claim 23, wherein said portion of the signal processing unit (219 comprises in addition a pre-amplifier (32).

25. The receiver device of one of claims 22 to 24, wherein the said output interface (24, 74, 76A, 76B, 76C) is formed by a plug member (74) comprising pins (76A, 76B, 76C), and wherein the printed circuit board is capable of being connected in the folded state to the plug member prior to being mounted within the housing.

26. The receiver device of claim 25, wherein said integral electronic unit (54) comprises contacts (75) for the output interface which are capable of engaging with the inner ends of the pins (76A, 76B, 76C) upon assembly of the integral electronic unit at the plug member (74).

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27. A method for manufacturing the receiver device (10) of claim 1, comprising: forming the loop like-conductor (56) on the flexible insolating substrate (55) while the substrate is in a planar condition, bending the first (58) and the second portion (60) relative to each other such that the first plane and the second plane become oriented at an angle of 60 to 120 degrees relative to each other and fixing the first and second portion in that orientation to each other, and mounting the antenna (20) and the signal processing unit (21) within the housing (64).

28. The method of claim 27, wherein said first (58) and second portion (60) of the antenna (20) are fixed in the bent condition at a carrier unit (70) of a plug member (74) forming the output interface (24) prior to being mounted together with the plug member within the housing (64).

29. The method of one of claims 27 and 28, comprising: electrically connecting the antenna (20) and the signal processing unit (21) prior to mounting the antenna and the signal processing unit within the housing (18, 52, 64) and trimming the antenna (20) and the signal processing unit (21) prior to mounting the antenna and the signal processing unit within the housing.

30. A method for manufacturing the receiver device of claim 4, comprising: forming the housing (54, 80) with the integrated antenna (20), and mounting the signal processing unit (21) within the housing.

31. The method of claim 30, wherein the antenna (20) is integrated into the housing (54, 80) by a molded interconnect devices (MID) technique.

32. The method of claim 31, wherein the housing (54, 80) is shaped first and subsequently the antenna (20) is integrated into the housing by modifying the surface of the housing.

33. The method of claim 32, wherein the plastic material is capable of being made conductive by laser activation and wherein the loop-like conductor (56) is created by laser activation of a surface portion of the plastic material of the housing (54, 80), followed by electroplating of the laser-activated surface portion in order to thicken the laser-activated surface portion.

34. The method of claim 32, wherein the loop-like conductor (56) is created by metal deposition from a metal evaporation source through a shadow mask onto a surface portion of the plastic material of the housing (54, 80).

35. The method of claim 32, wherein the loop-like conductor (56) is created by coating at least a portion of the surface of the housing (54, 80) with a metal layer, followed by selectively structuring the metal layer in order to create a loop-like metal structure.

36. The method of claim 35, wherein the metal layer is structured by selectively removing the metal layer.

37. The method of claim 31, wherein the antenna (20) is integrated into the housing (54, 80) during shaping of the housing.

38. The method of claim 37, wherein the housing (54, 80) is shaped by injection molding in a molding tool, wherein the loop-like conductor (56) is inserted into the molding tool and is overmolded in the molding tool.

39. Use of a receiver device (10) for receiving audio signals from a remote source (12), said device comprising: an antenna (120) for receiving radio frequency signals carrying audio signals, which antenna is either planar, thereby defining an antenna plane (92), or has an axial symmetry, thereby defining an antenna direction (92), a signal processing unit (21) for reproducing audio signals from the radio frequency signals received by the antenna, an output interface (24, 74, 76A, 76B, 76C) which is mechanically connected to an input interface (26) of a hearing instrument (16) worn at a user's ear (98) in order to supply the audio signals from the signal processing unit as input to the hearing instrument, and a housing (18, 64) enclosing the antenna and the signal processing unit, wherein said antenna plane or antenna direction is oriented at an angle of 30 to 60 degrees with respect to the direction (94) of the user's nose (96).

40. A method for manufacturing the receiver device of claim 22, wherein said printed circuit board (53) is manufactured including the loop-like conductor (56) of the antenna (20), electric components of the integral electronic unit (54) are mounted at the printed circuit board, the printed circuit board is brought into the folded state, and the printed circuit board is mounted in the folded state within the housing (64).

41. The method of claim 40, wherein the printed circuit board (53) is mounted in the folded state at a carrier unit (70) of a plug member (74) forming the output interface (24) prior to being mounted together with the plug member within the housing (64).

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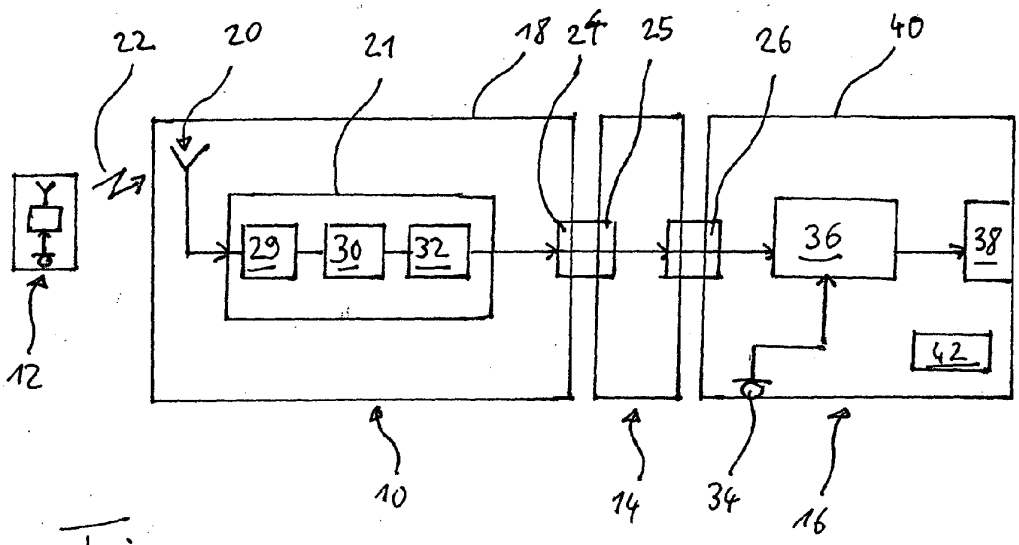

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Fig. 1

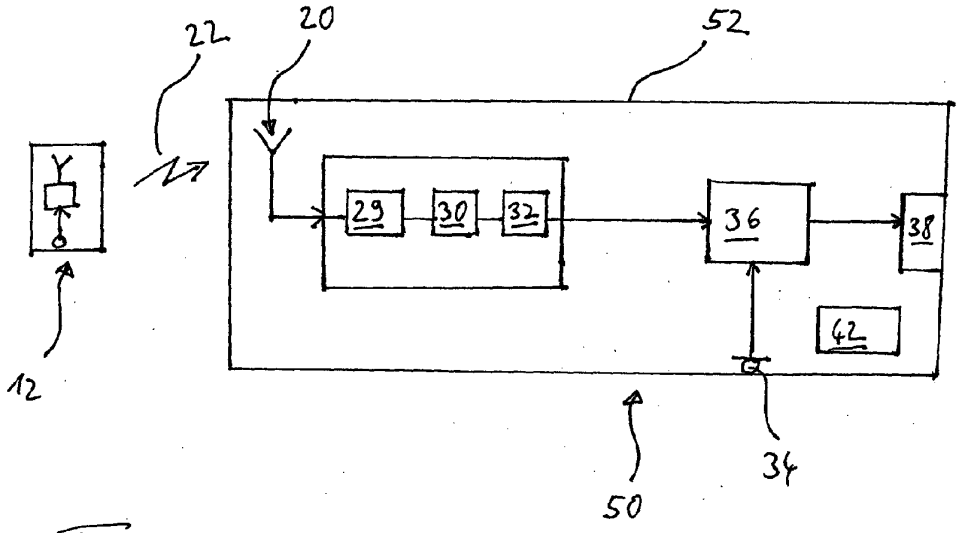

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Fig. 2

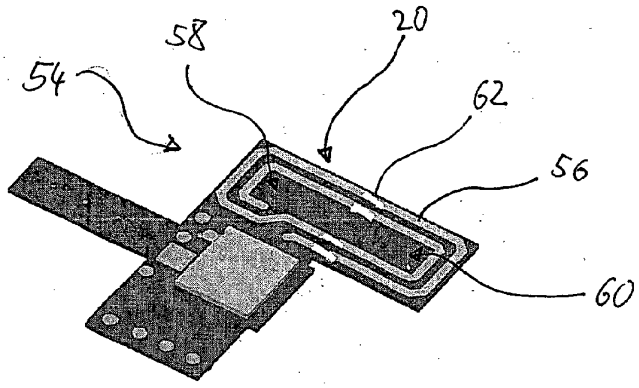
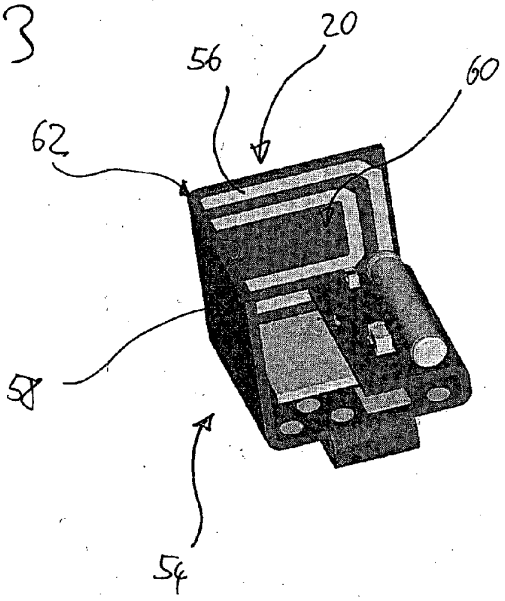


Fig. 3



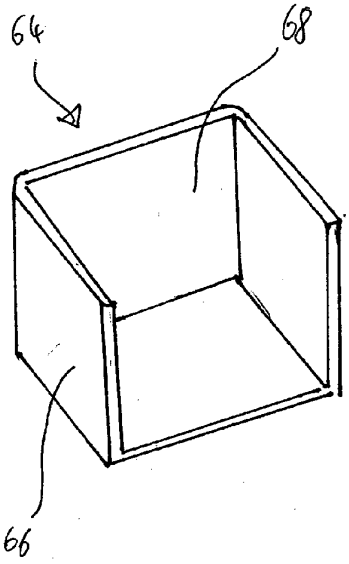


Fig. 4

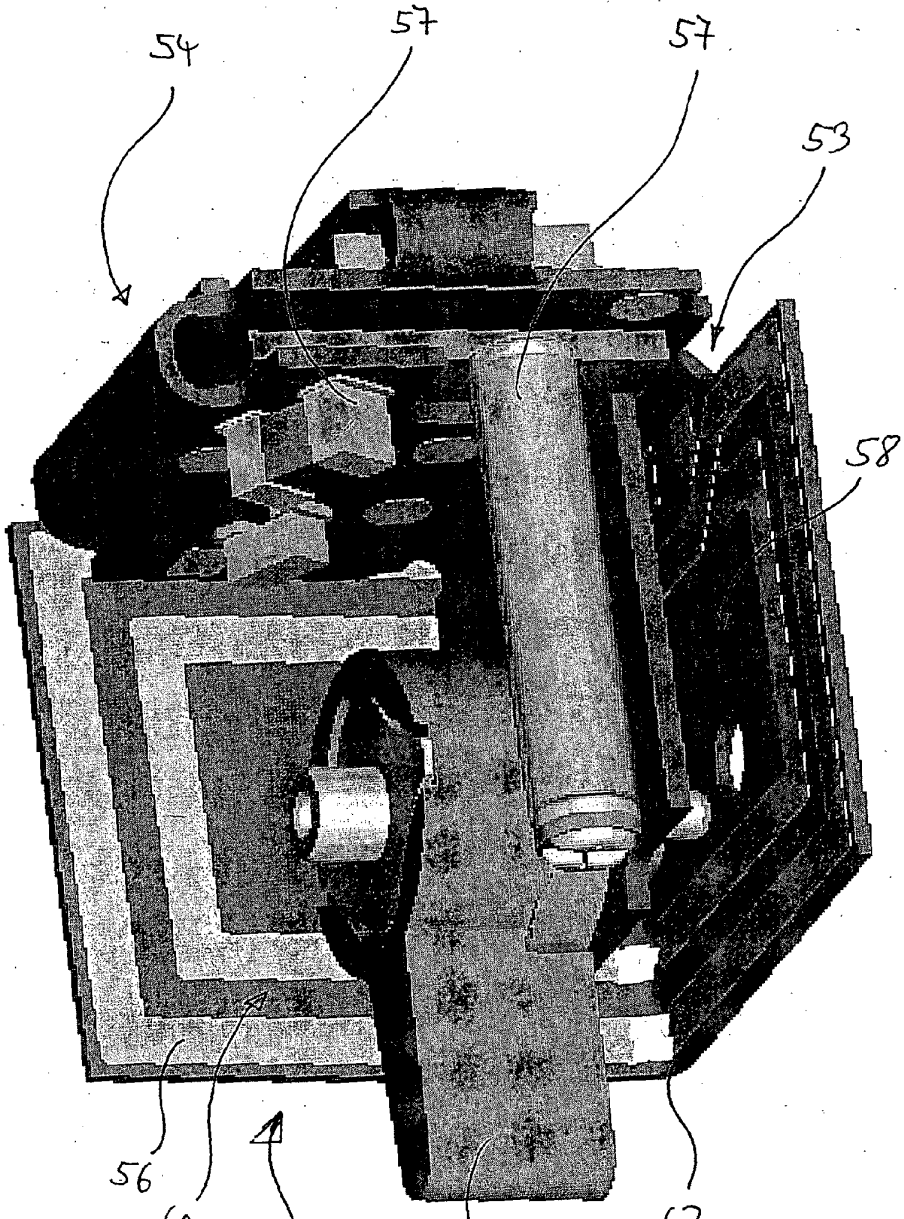


Fig. 5

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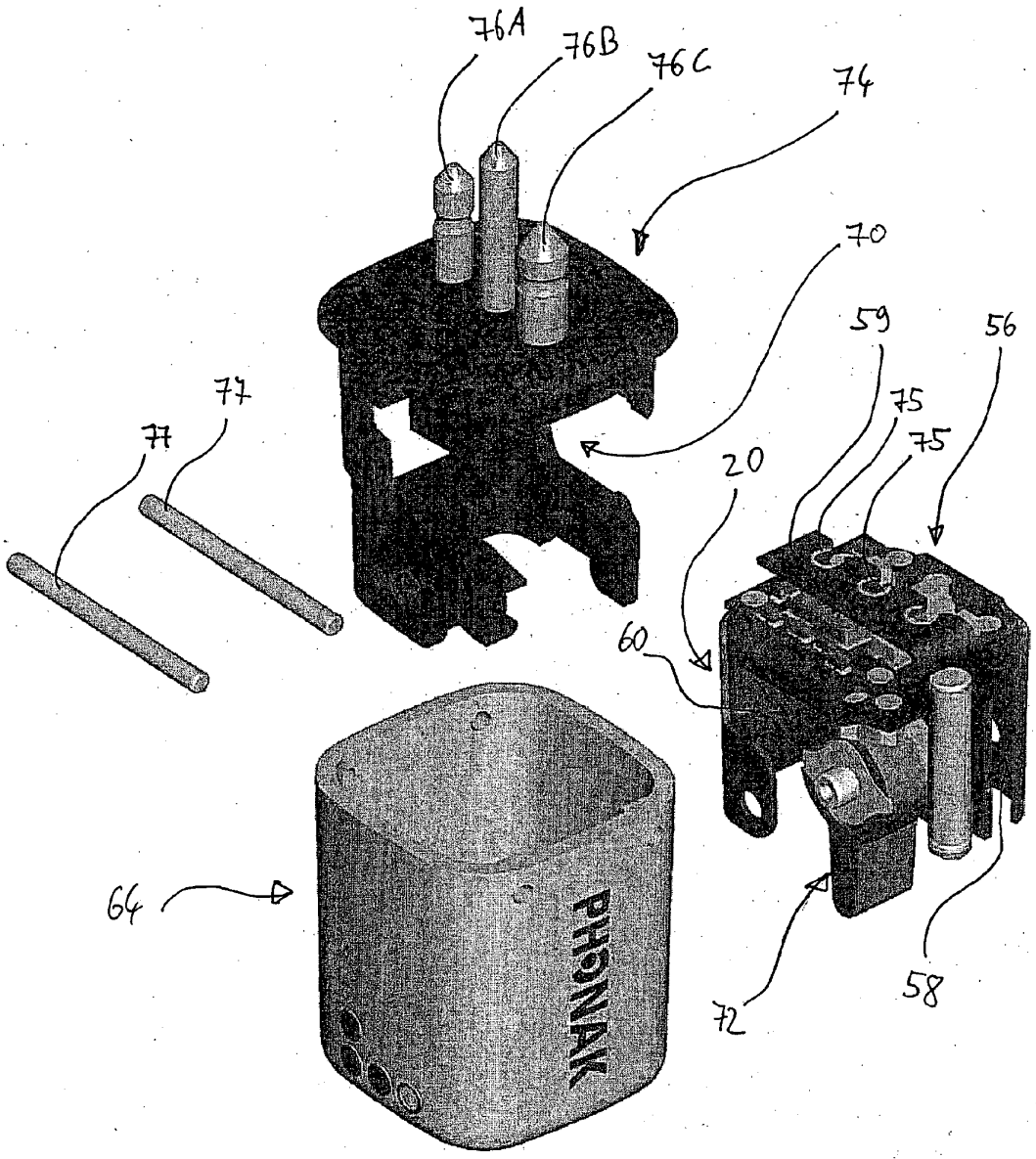


Fig. 6

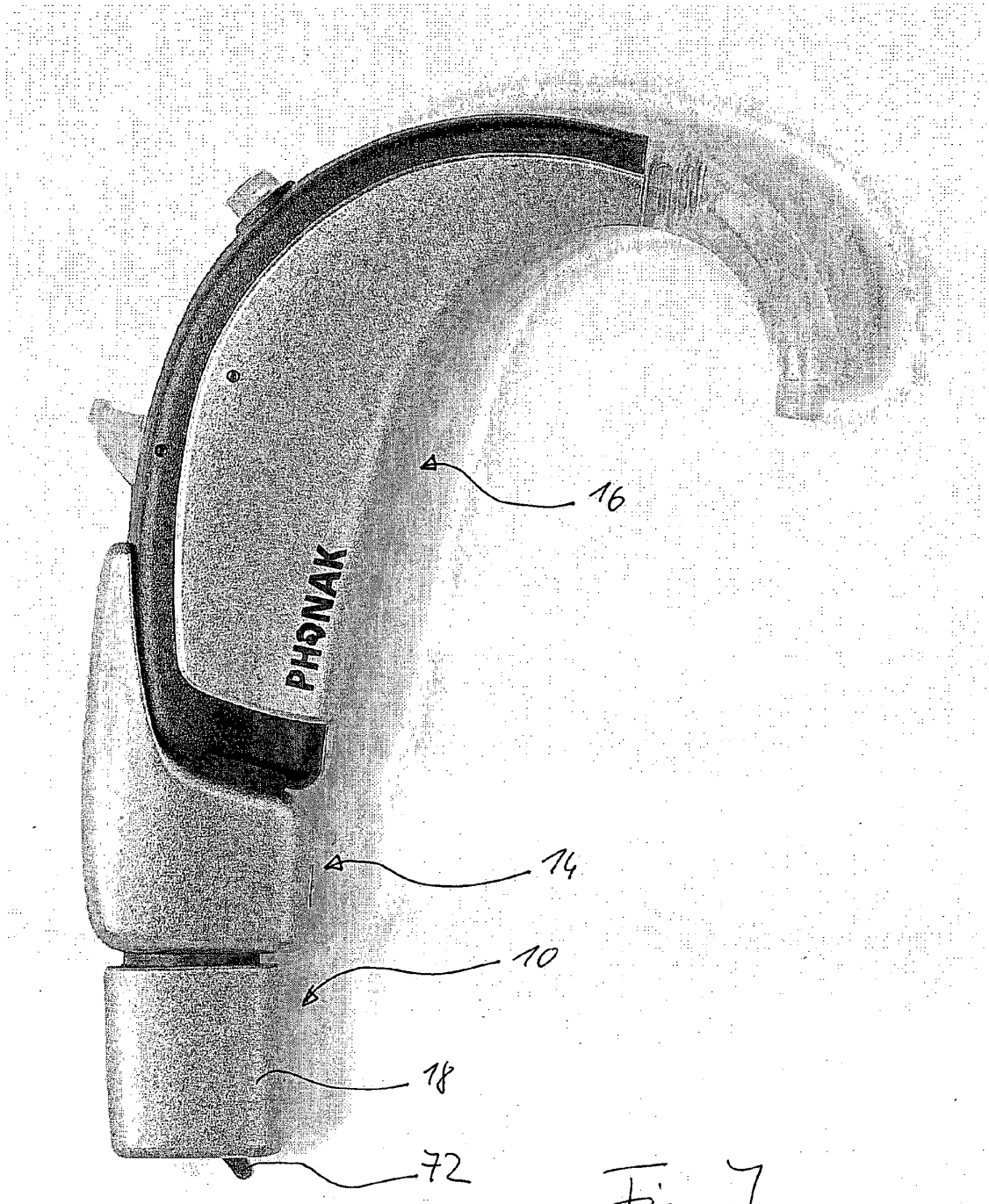


Fig. 7

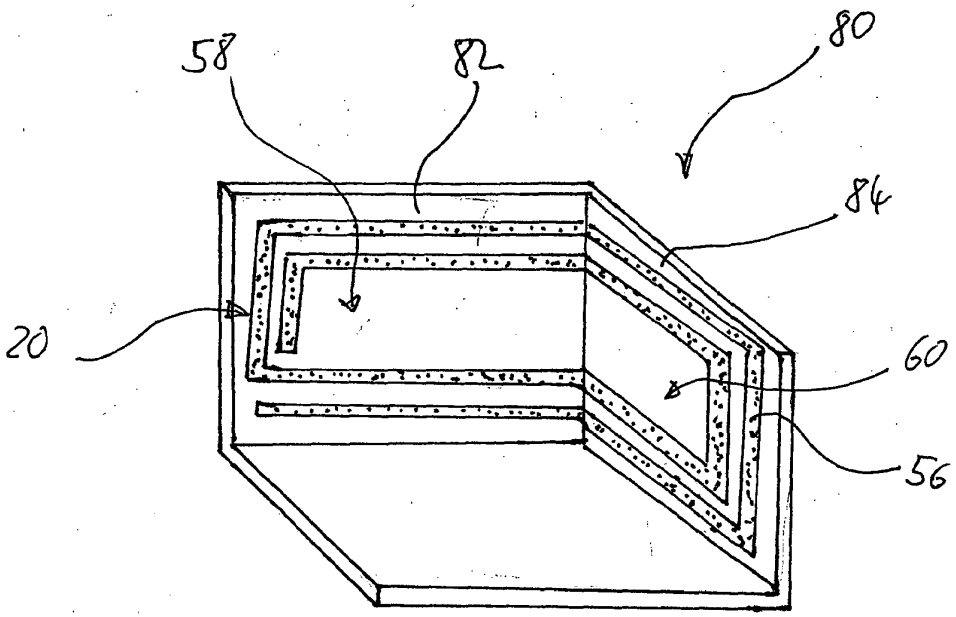


Fig. 8

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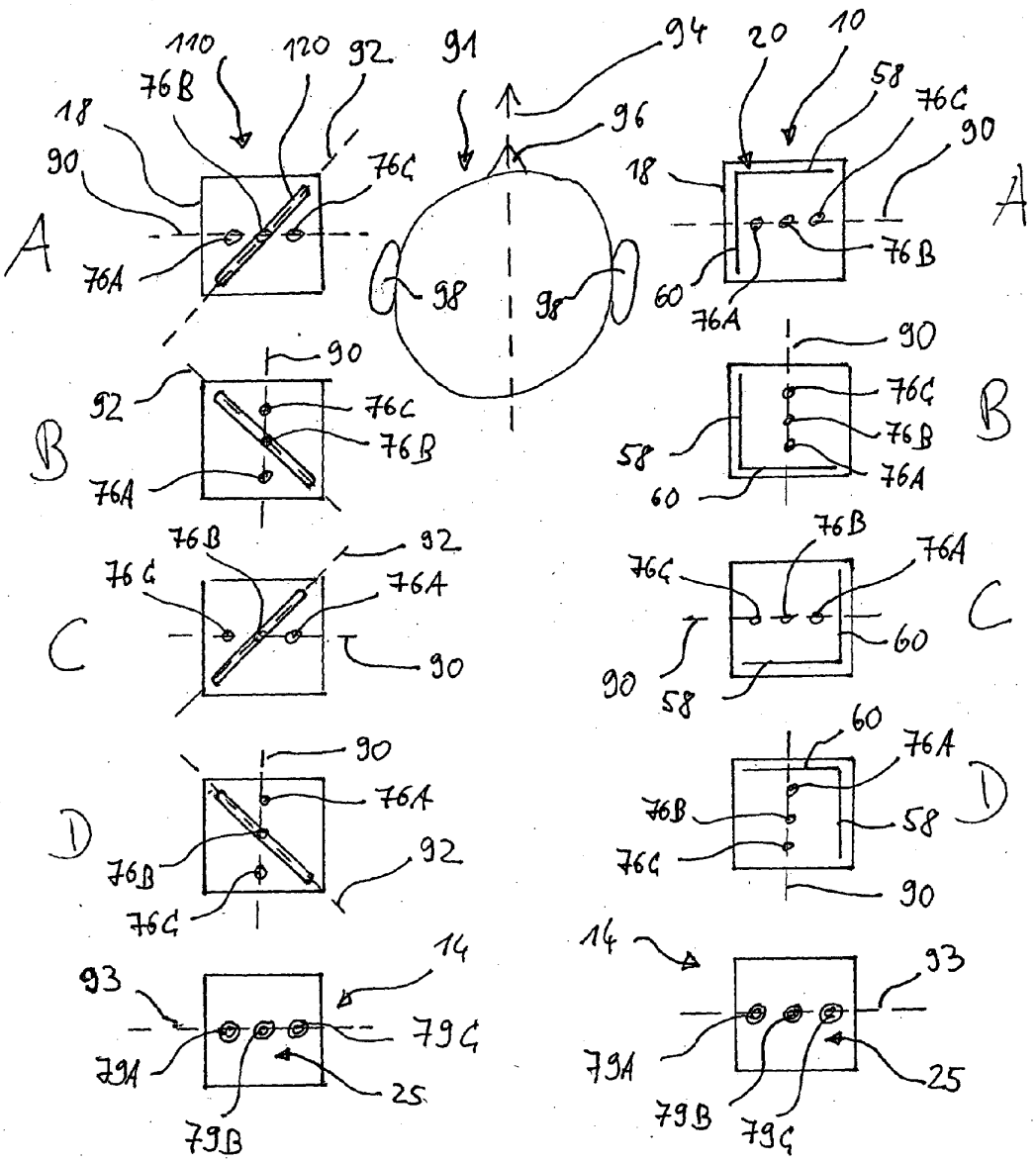

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Fig. 9

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(54) Binaural hearing device and method for controlling a such a hearing device

(57) A binaural hearing device system comprises a reception device (1) for one ear with at least two input acoustical/electrical converters(3a, 3b). Via a communication link (5) a signal (A_1) which depends on both input converter's output signals is transmitted to a second device (7) for the other ear which comprises at least an output electrical/mechanical converter.

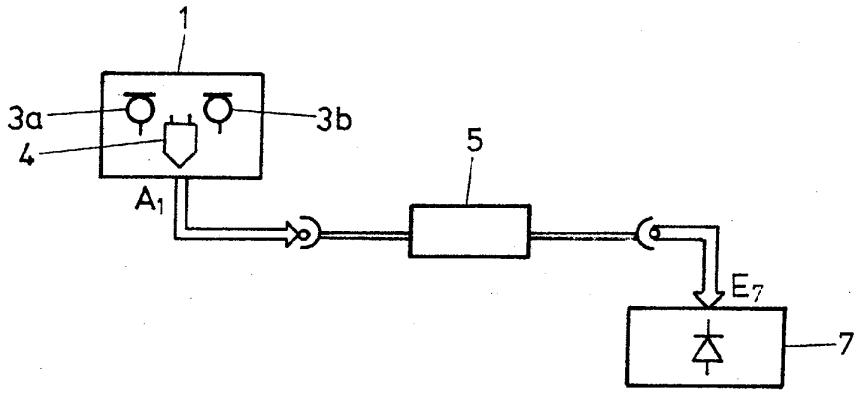


FIG.1

EP 1 320 281 A2

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Description

[0001] The present invention is most generically directed on binaural hearing device systems which necessitate a communication link between a device arranged in or a adjacent one ear and a device in or adjacent the other ear of an individual. The one-ear device comprises at least an arrangement of input acoustical/mechanical converters whereas the other ear device at least comprises an output electrical/mechanical converter.

[0002] From the WO 99/43185 such a binaural hearing device system is known, whereat each device associated to an ear comprises an input acoustical/electrical converter and an output electrical/mechanical converter. There is further provided a communication link between the two devices whereby data or signals are cross communicated via such link which are respectively dependent from the output signals of the respectively provided acoustical/electrical input converters. Thereby before the respective converter output signals are applied to the communication link they are analogue/digital converted whereby there may be implemented in the respective analogue/digital converters some additional signal preprocessing.

[0003] Today's monaural hearing devices customarily have at least two input acoustical/electrical converters for beamforming purposes. The binaural system according to the WO 99/43185 may be tailored to provide beamforming by using the two input converters provided at the respective one ear attributed devices. Thereby, as outlined above, data are cross-transmitted via the communication link which are possibly preprocessed but which comprise substantially more information than really needed. Further beamforming with two input converters placed one on each side of individuals head may be quite complex and inaccurate e.g. due to the head-related acoustical transfer functions HRTF which describe the effects of acoustical signals being "shadowed" by individuals head. Such shadowing occurs, dependent on direction of arrival of acoustical signals, asymmetrically with respect to both ears which on one hand allows spatial perception, on the other hand renders beamforming quite complex.

[0004] It as an object of the present invention to provide a binaural hearing device system and respectively a method for controlling such hearing device system whereat the technique of providing at least two input acoustical/electrical converters at one ear's device is maintained as known from monaural devices and additionally there is nevertheless applied to the communication link only one signal or data which is thereby dependent from the output signals of both of the at least two input converters at one ear's device. Thereby a significantly reduced amount of data is transmitted via said link compared with a case where, following the concept of the WO 99/43185, output signals of each input converter are separately transmitted via the link.

[0005] This object is resolved by the binaural hearing device system according to the present invention which comprises a first device for one ear of an individual, a second device for the other ear, a data/signal communication link between the first and the second device whereby the first device comprises at least a reception unit with at least two input acoustical/electrical converters and a signal processing unit the inputs of which being operationally connected to the electrical outputs of the at least two converters and which generates at a combined output a signal which is dependent on signals at both the said inputs whereby the signal link is provided at the output side of such processing unit and transmits data signals which depend upon the output signal of the processing unit whereby the second device comprises at least an output electrical/mechanical converter.

[0006] As is known to the skilled artisan there exist so called Complete-In-the-Channel, CIC-hearing devices whereat, due to complete introduction in the ear channel only one input acoustical/electrical converter is provided. Thereby whenever instead of the device mentioned above with at least two input converters, a CIC with only one input acoustical/electrical converter is to be applied according to the present invention's general concept, significant information and data reduction is achieved before transmitting data to the communication link, in that there is provided between the output of the one input converter and the communication link, a Wiener-Filter.

[0007] As was mentioned above the system according to the present invention provides in one embodiment the first device to be applied to one ear not having an electrical/mechanical output converter and thus only having in a reception unit the at least two acoustical/electrical input converters. This embodiment might be most valid e.g. if on any reason it is not possible to apply a device with at least two input converters at that ear where hearing shall be improved.

[0008] Thereby the second device does not comprise an input acoustical/electrical converter irrespective whether the first device has an output converter or not.

[0009] In a further preferred embodiment an output electrical/mechanical converter provided at the first device is operationally connected to the output of the processing unit and is thus driven by a combined signal or data dependent on both outputs of the at least two input acoustical/electrical converters provided.

[0010] In a still further preferred embodiment the system according to the present invention has the reception unit of the first device as a first reception unit whereby the at least two input acoustical/electrical converters thereat are first acoustical/electrical converters. Additionally the signal processing unit still at the first device is a first signal processing unit.

[0011] Further the output electrical/mechanical converter at the second device is considered as a second output electrical/mechanical converter. The first device comprises a first output electrical/mechanical converter

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and the second device a second reception unit.

[0012] Thus both devices for each of the two ears have respective reception units and thus input acoustical/electrical converters and respective output electrical/mechanical converters.

[0013] Nevertheless the second reception unit at the second device needs not necessarily have more than one input acoustical/electrical converter although providing also there at least two input acoustical/electrical converters is preferred.

[0014] Further the communication link which is provided in all embodiments according to the present invention, for communicating between devices adjacent or in the respective ears, maybe wirebound and/or based on optical fiber and/or on wireless communication.

[0015] Whenever both ears devices are equipped with input acoustical/electrical converters in a preferred embodiment both devices are equipped with at least two of such converters which gives the possibility to provide at both devices beamforming ability. Thereby further preferably also the second reception unit is equipped with a signal processing unit whereby, further preferred, the inputs of such processing unit are operationally connected to the electrical outputs of the second input converters at the second reception unit. This processing unit generates at a respectively second output a signal which is dependent on signals at both said inputs of the second signal processing unit whereby the signal link is provided at the output side of the second signal processing unit. Thus via the addressed signal or communication link combined signals dependent respectively on the output signal of at least two input converters are bidirectionally transmitted from one device to the other and vice versa.

[0016] Thereby and in a further preferred mode or embodiment the output of the first signal processing unit is operationally connected to a first input of a weighting unit and the output of a second signal processing unit is operationally connected to a second input of the weighting unit. The weighting unit has a first output which is operationally connected to an input of a first output converter and has a second output which is operationally connected to the input of the second output converter. Thereby the weighting unit may be construed decentralised e.g. in both devices. The weighting unit has a control input and varies operational connection or signal transfer between the first input and the first output, the first input and the second output, the second input and the first output and finally the second input and the second output. Such signal transfers are controlled by a signal or data applied to the control input of said weighting unit. Thereby such operational connections between respective inputs and outputs are formed preferably frequency or frequency-band specifically and the respective functions which are controlled independently from one another are possibly but not necessarily complex functions.

[0017] So as to determine how the operational connections between respective inputs and outputs at the weighting unit have to be controlled, especially according to the acoustical surrounding present, the control input of the weighting unit is preferably connected to an output of a classification unit which later has at least one input operationally connected to an output of at least one of the reception units.

[0018] In a further most preferred embodiment the first device comprises a beamformer unit which has a beamcontrol input and an output. Via the beamcontrol input the directional characteristic of the beam as an amplification characteristic in dependency of spatial angle at which an acoustical signal impinges on the device, may be varied.

[0019] There is further provided a detection unit for detecting the direction of arrival of an acoustical signal which impinges upon the reception unit which unit generates at an output an output signal in dependency of said direction of arrival. This output is operationally connected to the beamcontrol input of the beamformer unit so that e.g,. a source of acoustical signal the direction of arrival of which having been detected may be more accurately tracked by accordingly directing a maximum amplification direction of the beam upon such a source. Accordingly a source, as e.g. a noise source, the direction thereof having been detected may be cancelled by controlling the beam so that it establishes in that noise source direction minimum amplification.

[0020] As was mentioned above in a preferred embodiment there is provided a weighting unit whereat signal transmission between respective inputs and outputs is controlled. Thereby control of such signal transmission is made dependent from the result achieved in a classification unit the input thereof being operationally connected to at least one output of at least one of the reception units.

[0021] Departing from this embodiment and in a further preferred mode there is provided at the system a determination unit for the direction of arrival of an acoustical signal impinging on at least one of the devices whereby such direction determination unit is interconnected between at least one input of the classification unit and at least one output of at least one of the reception units at the devices.

[0022] Thus the classification which finally controls signal transfer at the weighting unit at least comprises classification of signals which depend on direction of arrival. Thereby and as a further improvement of such embodiment there is provided at least one histogram forming unit, the input thereof being operationally connected to at least one output of at least one of the reception units. The output thereof is operationally connected to an input of the classification unit. Thus classification at least comprises classification based on a histogram result. Most preferably and with an eye on providing a direction of arrival determination unit such histogram forming unit is provided with an input operationally con-

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nected to an output of the determination unit and an output operationally connected to the classification unit. Thereby classification at least comprises classification of a histogram function of a signal or of signals which identify such direction of arrival.

[0023] The object mentioned above still further is resolved by the method for controlling a hearing device system which comprises at least a reception unit at a first device for one ear which has at least two inputs acoustical/electrical converters and at least an output electrical/mechanical converter at a second device for the other ear and a communication link between the first and the second device which method comprises the steps of generating in dependency of output signals of the at least two input converters a combined signal and transmitting such combined signal via the communication link.

[0024] For applying the method according to the present invention to CIC hearing devices the method according to the invention comprises providing instead of the at least two input converters only one converter and construing the first device as a device to be completely introduced into the ear channel and further comprises a step to treat the output of the one input converter by a Wiener-Filter and transmitting signals dependent from the output of the Wiener-Filter via the communication link.

[0025] The present invention and the object thereof is further resolved by the method for producing a drive signal for a electrical/mechanical output converter of a binaural hearing device which method comprises the steps of acoustical/electrical converting impinging acoustical signals at at least two input converters of a device to be applied adjacent individuals one ear, transmitting a combined signal dependent from both said convertings via a link to a further device to be applied adjacent or in individuals other ear and generating the drive signal in dependency of the transmitted signal.

[0026] Further preferred embodiments of the methods according to the present invention as well as of the system according to the present invention will become apparent to the skilled artisan when reading the following description of preferred embodiments of the present invention as well as the claims.

[0027] The present invention will now be further described with the help of figures. They show examples of preferred embodiments, namely:

Fig. 1 By a schematic, simplified functional-block/signal-flow representation, a first embodiment of the system according to the present invention and operated according the methods of the present invention;

Fig. 2 in a representation form in analogy to that of fig. 1 a further embodiment of the present invention;

Fig. 3 again in a simplified schematic functional-block/signal-flow representation a still further embodiment according to the present invention again operating according to the methods of the present invention;

Fig. 4 still in the same representation form a further embodiment of the present invention;

Fig. 5 by means of a simplified schematic functional-block/signal-flow representation a subembodiment for automatic beamcontrol e.g. to track acoustical sources and/or to cancel reception of acoustical sources. Such embodiment may preferably be incorporated within the embodiments according to the present invention;

Fig. 6 departing from a system or methods according to fig. 4 still in a simplified schematic functional-block/signal-flow representation an improved embodiment of such system or methods;

Fig. 7 by means of a simplified schematic functional-block/signal-flow representation a system or method for controlling a hearing device as a function of direction of arrival of acoustical signals as detected and preferably classified;

Fig. 8 examples of direction of arrival behaviours as appearing on a histogram function to explain some of more simple classification criteria as preferably exploited at the system or methods of fig. 7 as well as at systems or methods to be shown with the help of the fig. 9 and 10;

Fig. 9 in form of a simplified schematic functional-block/signal-flow representation an improved and today preferred form of an embodiment of the system according to the present invention and of the methods according to the present invention;

Fig. 10 departing from the representation of fig. 9 a more detailed representation of such system or methods making use of direction of arrival detection as described in more details in the WO 00/68703 which accords with the US application no 09/636 443 and 10/180 585.

[0028] According to fig. 1 a system according to the present invention operating according to the method of the present invention both under a first aspect thereof is schematically shown by means of a simplified functional block/signal flow diagram in a minimal configuration. There is provided an acoustical reception unit 1 with at least two acoustical/electrical converters 3a and 3b, both with a respective acoustical input and an elec-

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trical output. Reception unit 1 may incorporate e.g. respective analog to digital converters connected to the outputs of the converters 3a, 3b, time domain to frequency domain conversion units downstream such analog to digital converters and has a signal processing unit 4 for processing signals in dependency of the analog signals appearing at the outputs of the converters 3a, 3b. Processing unit 4 generates at an output A_1 of reception unit 1 a signal or data which is result of combined processing of signals dependent on the output signals of both converters 3a and 3b: The output signal at A_1 depends on the output signals of both converters 3a, 3b. This signal or data at output A_1 possibly further processed at respective signal processing units (not shown) generates a signal or data, which is dependent on the output signal or data at A_1, which is transmitted to a transmission link 5, which again may incorporate further signal processing. At the output side of transmission link 5 a signal or data, which is dependent on the signal appearing at the output A_1 of unit 1, is input to an input E_7 of an electrical/mechanical converter unit 7. Unit 1 is applied adjacent or within one of an individual's ears, unit 7 to the other.

[0029] The system as shown in fig. 1 is in a preferred embodiment a hearing aid system i.e. a therapeutical system. Unit 7 is thereby an outside-the-ear or an inside-the-ear converter unit or an implanted or implantable unit. By this minimal system acoustical signals are received on one of individual's ears and control hearing at the other ear. Such a system may be provided, where on any reasons, applying the reception unit 1 is not possible or difficult on that ear where hearing shall be improved or reinstalled.

[0030] The concept of applying a reception unit as of unit 1 at or adjacent one ear and transmitting signals or data dependent on the received acoustical signals at such, reception unit to the other ear for improving hearing at that other ear, this concept per se is considered inventive, irrespective of how reception unit, signal link to the other ear and a other's ear converter unit as of unit 7 of fig. 1 are conceived: Under this concept one ear is only provided with an electrical/mechanical unit and no reception unit. The embodiments of figs 1 to 3 clearly fall under such concept. In any case the link 5 may be electric wire based, optical fiber based or may be a wireless communication link.

[0031] The double-line arrows shown in Fig. 1 and following figures represent signal or data communication paths. Along such signal path additional signal processing by respective units may be established. The double-arrows may indicate a direct signal transmission, but rather stand for an operational connection, in which signals are transmitted and processed in direction of the arrow.

[0032] By the system according to fig. 1 only data or signals are transmitted via transmission link 5, which have been preprocessed as by combining signals of at least two acoustical to electrical input converters 3a, 3b.

[0033] In fig. 2 there is shown in a representation, in analogy to that of fig. 1, a second preferred embodiment, which only differs from that of fig. 1 in that unit 1 of fig. 1 is now conceived as a unit 10 to be applied completely introduced in an individual's ear channel, a so-called CIC-device. As known to the skilled artisan such a CIC unit customarily has only one input acoustical to electrical converter 3c. By means of a digital signal processing unit 11, which is operationally connected e.g. via time domain to frequency domain converter and analog to digital converter to the analog output of converter 3c, at least a Wiener-filtering is performed. The output signal or data of converter 3c is processed by a Wiener filter to result in significantly preprocessed data and perceptual information reduction thus enabling simpler source/channel coding before being transmitted via communication link 5 to the electrical to mechanical converter unit 7.

[0034] In fig. 3 there is shown in a representation in analogy to that of the figs. 1 or 2 a further preferred embodiment of the system according to the present invention, which operates according to the method of the present invention. According to the system of fig. 3, the difference to the system of fig. 1 is that the output A_1 of reception unit 1 is not only, via transmission link 5, operationally connected to the input E_7 of the electric/mechanic converter unit 7 at the other of individual's ears, but output A_1 is additionally operationally connected to an electrical/mechanical converter unit 7b, which is provided at the same ear as reception unit 1.

[0035] It is evident that in dependency of the signals or data at output A_1 the left ear and the right ear units 7a and 7b have normally to be differently operated. Thus there are generically installed different and/or differently operating signal processing units as on one hand between the output A_1 and link 5, link 5 and input E_{7a}, and on the other hand output A_1 and input E_{7b} of unit 7b. In the case of the embodiment of fig. 3 and as shown in dashed-pointed frame, the units 1 and 7b are preferably incorporated in a unitary hearing device, especially in a hearing aid device being a behind- or an in-the-ear hearing device.

[0036] Instead of providing a reception unit 1 with at least two input acoustical to electrical converters 3a and 3b as of fig. 3, this unit may be construed according to unit 10 of fig. 2, i.e. as a CIC-unit.

[0037] According to the embodiment of fig. 3 there is in fact established a MASTER-acoustical control by reception unit 1 at one ear of the individual, whereas a hearing device without an input acoustical to electrical converter unit is operated at the other ear as a SLAVE device.

[0038] Departing from the system and method as explained with the help of fig. 3 a further preferred embodiment of the invention under the first aspect thereof is shown in fig. 4, still in a representation in analogy to that of the figures 1 to 3.

[0039] According to the system of fig. 4 there is pro-

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vided for the left ear of an individual a reception unit 1_L and for the right ear a reception unit 1_R. Both reception units 1_L and 1_R are conceived with respect to signal or data processing as was explained with respect to reception unit 1 in context with fig. 1. Instead of units 1_R and 1_L being conceived according to unit 1 of fig. 1, one or both thereof may be conceived according to unit 10 of fig. 2. A signal or data dependent from the signal or data at the output A_{1L} of reception unit 1_L is fed to an input E_{9L} of a selection unit 9. A signal or data which is dependent from the signal or data appearing at the output A_{1R} of the right ear reception unit 1_R is fed to an input E_{9R} of the selection unit 9. There is further provided a left ear electrical/mechanical output converter unit 7_L and a right ear electrical/mechanical output converter unit 7_R.

[0040] The selection unit 9, as schematically shown by a switching arrangement, has an output A_{9L} and an output A_{9R} respectively operationally connected to the inputs of output converters 7_L, 7_R. Signals or data appearing at either of the outputs A_{9L} or A_{9R} may operationally be connected to both electrical to mechanical converter units 7_L and 7_R. Under the control of a selection-control unit 12 and, as schematically shown in unit 9 by an arrangement of switches, the input E_{9L} or the input E_{9R} is operationally connected to both of the converters 7_L, 7_R. Thereby, whenever the operational signal or data connection within selection unit 9 is established according to that switching position shown in fig. 4, both converters 7_L and 7_R are operationally connected to the right ear reception unit 1_R, and therefore the right ear reception unit 1_R is the MASTER. In analogy, unit 1_L becomes MASTER whenever the units 7_L and 7_R are operationally connected to the input E_{9L} of selection unit 9.

[0041] In this embodiment again the right ear units 1_R and 7_R are preferably incorporated in a unitary right ear hearing device, be it a hearing aid device or be it a hearing device for other than therapeutical appliances. In analogy the units 1_L and 7_L are incorporated in a respective left ear unitary device. Such hearing devices may thereby be in-the-ear or outside-the-ear hearing devices or their output converters 7_L and/or 7_R may be construed as implantable devices. Further, the right and left ear devices do not necessarily have to be of the same type, e.g. an in-the-ear and an outside-the-ear hearing device may be combined, an outside-the-ear and an implant device etc.

[0042] Looking back on fig. 3 it has been shown that the acoustical signal impinging on unit 1 at one ear, e. g. at the left year, binaurally controls both electrical to mechanical output converter units 7a and 7b. We have established that double-lined arrows stand for operational signal or data communication and not necessarily for direct connection. Thus, along operational connections processing as by processing units, especially DSP's, may be done. For example: As according to fig. 3 the acoustical signals impinging on unit 1 do control both output converters 7 and thus the head-related transfer function HRTF for the SLAVE side with converter 7a is lost, there will preferably be provided as shown in dashed line a DSP 13 exclusively influencing signals or data input to the SLAVE converter 7a and whereat the respective HRTF is taken into account. So as to properly set the parameters of processing in DSP unit 13 for taking the HRTF functions into account, the reception unit 1 detects direction of arrival DOA as denoted by φ in fig. 3 and there will be transmitted additionally to the signal or data dependent from those appearing at output A_1 of unit 1, via link 5, a DOA-significant signal or data to DSP 13 as shown by signal DOA. Further, there will be preferably provided a DSP 14 just upstream the input E_{7b} and DSP 13 or a further DSP to input E_{7a} as well as DSP 14 will take in account different signal processing needs according to the hearing improvement needs at the respective ears.

[0043] When looking to the embodiment of fig. 4 in analogy to the just given explanations with respect to the system of fig. 3, whenever the right ear device is MASTER, the HRTF will preferably be considered for the left ear converter 7_L, i.e. the SLAVE and vice versa. Thus, the left ear HRTF is taken into account by a DSP 16, and the right ear HRTF by a DSP 18. Preferably that one of the units 1L and 1R, which acts as a MASTER, provides for data about direction of arrival DOA (not shown) so as to control the transfer characteristic of the respective HRTF DSP 16 and 18.

[0044] With an eye on Fig. 1 or 2, there the processing unit 4 will preferably take the HRTF of the left side ear into consideration.

[0045] With respect to one preferred possibility for detecting direction of arrival DOA of acoustical signals at the reception units 1, 10, 1L and 1R, we refer to the WO 00/68703 "Method for localizing direction" of the same applicant, wherein a technique for detecting such direction of arrival DOA is completely disclosed, and which shall be incorporated with respect to DOA detection into the present description. This WO 00/68703 accords with US application no. 09/636 443 and no. 10/180 585. Thereby, the reception units 1, 1L, 1R may preferably further comprise beam formers as are e.g. described in the WO 00/54553, according to US application No. 09/267 742, the WO 99/04598, according to US application No. 09/146 784, the WO 99/09786, according to US application No. 09/168 184, all of the same applicant.

[0046] Thus, in one preferred embodiment such units 1, 1L, 1R provide for both, namely beam forming as well as detection of DOA. Thereby, in a further preferred embodiment beamforming is controlled by the DOA.

[0047] This preferred form of realizing the reception units 1, 1L, 1R as discussed up to now is schematically shown in fig. 5. Thereby, the units 1, 1L, 1R comprise a beamforming subunit 20 with at least two input acoustical/electrical converters. At the output of such unit, which accords to output A_1 or A_{1L}, A_{1R} there appear electrical data or signals in dependency of acoustical

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signals impinging on the at least two input converters and amplified according to a predetermined characteristic in dependency of spatial angle with which the acoustical signals impinge on the input converters. The outputs of the acoustical to electrical converter are further exploited e.g. according to the teaching of the WO 00/68703 so as to provide for a signal which is indicative of the direction of arrival DOA of the acoustic signals. Thereby preferably and as described in the said WO 00/68703, there is performed a histogram of the DOA signals, as will be discussed later. The output of a histogram-forming and evaluating unit 22 controls beamformer unit 20 at a control input C_{20} e.g. to track an acoustical source selected with high amplification or to delete such acoustical source by low amplification.

[0048] Turning back to the system of fig. 4, it may be seen that the data link 5, which was shown in the figs. 1 to 3, has not been shown anymore. Such data link, by which signals or data are or is transmitted from one ear side to the other, may be provided in the system as of fig. 5, wherever felt best. The selection unit 9 may e.g. be incorporated in one of the left ear or right ear devices, e.g. in the left ear device and then the addressed data link 5 will be provided at 5' as shown in fig. 5. On the other hand the selection unit 9 may be split into left ear device- and right ear device-units, and then the data link 5 would be established and following the representation of fig. 4 practically within selection unit 9.

[0049] Further, with an eye on fig. 4, this system clearly operates one of the two devices as a MASTER, the other one, and thereby especially the output converter 7 thereof, as a SLAVE. Changing this MASTER/SLAVE relation occurs abruptly and it is not possible to gently control the MASTER/SLAVE weighting of the two devices. This becomes possible by the improvement on fig. 4, which shall be explained with the help of fig. 6.

[0050] According to fig. 6, wherein units which correspond to units already described in context with fig. 4 have been denoted with the same reference number, the selection unit 9_w in fact is a weighting unit. Therein, the influence of a signal or data dependent from such signal or data at output A_{1L} upon signal or data respectively appearing at the outputs A_{9L} and A_{9R} is continuously adjustable, as shown schematically by variable coefficients α, β. In analogy the influence from output A_{1R} upon the two outputs A_{9L} and A_{9R} of unit 9_w is adjusted as schematically shown by variably controllable coefficients γ and δ. The coefficients α, β, γ, δ are preferably frequency dependent or at least dependent from frequency bands and are possibly of complex value. These weighting coefficients are controlled by a selection control unit 12_w.

[0051] In the embodiments according to the figs. 5 and 6 there is provided respectively a selection control unit 12 or 12_w not having been described yet. The selection control unit 12 and respectively 12_w are in fact classification units, whereat the instantaneously prevailing acoustical environment and/or the time development in the past up to the present of such acoustical surrounding and even a trend estimation for future development of such acoustical signals is classified according to predetermined criteria as e.g. disclosed in the WO 02/32208 which accords with US application no. 10/059 059 or in the WO 01/20965 according to US application no. 2002-0 037 087 or in the WO 01/22790 according to US application no. 2002-0 090 098. In any case to the classifier and control units 12, 12_w there is input information about the acoustical signals received at units 1, 1_L and/or 1_R as shown at 13 in Fig. 4, at 13a, 13b in Fig. 6. Under a second aspect of the present invention a preferred classification technique shall be described in the following, which is most apt to be combined with the present invention under its first aspect described up to now.

[0052] This second aspect of the invention is schematically shown in fig. 7, by a representation in analogy to that used throughout the figs. 1 to 6. It comprises a reception unit 30 with at least two input acoustical to electrical converters. The unit 30 operates so as to generate an output electrical signal or data at output A_{30} indicative of the spatial direction of arrival DOA with which an acoustical signal impinges upon the acoustical inputs of the input converters 31a and 31b as provided. Such a unit is known e.g. from the WO 00/68703 which accords with the US application No. 09/636 443 and 10/180 585 of the same applicant. From the instantaneously monitored DOA there is generated by means of a processing unit 32 a histogram function of DOA. This is also known from the WO 00/68703. Thus, under the second aspect of the invention there is formed a histogram of the instantaneously prevailing DOA. According to the second aspect of the invention it is the DOA-histogram which is used as entity for classifying the acoustical signals in unit 34, which impinge upon the unit 30 and for controlling system adjustment especially according to figs. 4, 5, or 6. Thereby and as schematically shown in fig. 7 by dashed-dotted lines, the reception unit 30 is preferably a part of a hearing device system 36. The signals or data representing audio signals are generated by unit 30 at output A_{230}, if that unit 30 performs combined tasks of DOA detection and audio signal processing. The histogram generated at unit 32 is now classified in classifying unit 34, which controls at its output most generically the behavior of a hearing device system, be it a monaural system, but most preferably of a binaural hearing device system as shown in Figs. 1 to 6.

[0053] Accordingly in fig. 8 there is shown more than one output of classifying unit 34 representing different controls to the hearing device system according to different types of histogram appearance and thus of acoustical source behavior in the acoustical surrounding U of fig. 7 of the hearing device system, and thus of an individual carrying such system.

[0054] In fig. 8a there is shown purely as an example such a histogram function represented by the overall time or in fact the overall number n of measuring sam-

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ples, which result in a specific DOA spatial angle φ. For the $DOA_{\varphi 0}$ a relatively sharp peak is present indicating that at that angle φ_0 to the acoustical input of the converters 31a and 31b there is a significant acoustical source in the acoustical surrounding U. At φ_1 there is a second yet less relevant acoustical source present in the surrounding U.

[0055] Departing from this histogram (a) some possible evaluations in time shall be discussed. According to fig. 8(b) at the DOA φ_0 the peak has become broadened and its amplitude has dropped. This means e.g. that the acoustical source at the angle φ_0 has become diffuse, which may be caused by an increase of distance between the reception unit 30 and the acoustical source in the surrounding U. According to fig. 8(c) and still considered as an evolution in time of the situation as present according to fig. 8(a), it may be seen that the histogram has been shifted by an angle Δ. This means that the reception unit 30 has rotated relative to the acoustical surrounding U, in other words that the individual carrying a system with unit 30 has turned his head by the angle Δ. This is identified because the relative positioning of the sources in the surrounding U according to fig. 8(a) at φ_0 and at φ_1 remains stable.

[0056] According to fig. 8(d) the peak appearing at the DOA φ_0 according to fig. (a) now appears at a different angle φ_2, whereas the source of at φ_1 according to fig. (a) still appears at the unchanged angle φ_1. This means that the source at φ_0 according to fig. (a) has moved to the new angular position φ_2, whereby the reception unit 30 has not rotated, i.e. the individual has kept his head stationary. From these explanations it may be seen which kind of criteria are used in classifying unit 34 of fig. 8 to establish a relevant acoustical source, increasing distance, decreasing relevancy of a source, appearance/disappearance of a source movement of individual's head relative to the acoustical surrounding, angular movement of a source in the surrounding U, etc.

[0057] From combining and adding further classifying criteria an intelligent evaluation of the acoustical surrounding is performed and by the respective results the behavior of the hearing device system 34 is controlled. This may include source tracking by controlling beamforming and/or with an eye back on fig. 5 and 7 appropriate distribution of the influence or signal transfer of binaurally provided reception units upon binaurally provided output converters.

[0058] Thus under the second aspect the present invention is directed on classifying signals or data which are indicative of the DOA and controlling the status or behavior of a hearing device, be it a monaural or binaural device in dependency of the classification result. Thereby most preferably classification is performed upon on data or signals wherefrom a histogram has been formed.

[0059] In fig. 9 there is shown a preferred embodiment, which combines the invention under its first aspect realized as was explained with the help of fig. 6 and under its second aspect.

[0060] A left ear reception unit 40_L of a left ear hearing device is conceived as a beamformer with at least two input converters 41_L. The right ear hearing device, as an example, is equally construed as the left ear device and thus comprises a reception unit 40R equal to the unit 40_L. In analogy to the representation in fig. 6 at the respective outputs A_{1L}, A_{1R} electrical signals or data are generated as a result of processing the output signals of the converters 41. These signals are thus dependent on the acoustical signal impinging on the reception units, amplified according to the beamformer characteristics. The units 40 preferably comprise a respective beamformer control input BFC_L and BFCR, by which the shape of the beamformer characteristic, but especially the angle θ of maximal amplification may be adjusted. The units 40 further generate output signals, which are indicative of the DOAφ of acoustical signals impinging on the acoustical inputs at the units 40. Signals or data dependent from these output signals DOA_L, DOA_R are respectively input to histogram-forming units 44_L, 44_R. The units 40 combined with histogram-forming units 44 may and are preferably realized as described in the WO 00/68703, which accords with the US application No. 09/636 443. Thereby and as seen in this paper the beamformers are based on the delay-and-add/subtract principal and thus the beamformer control input BFC_L and BFC_R may e.g. adjust the delay τ. It is well-known to the skilled artisan that by establishing and varying the delay τ in a delay-and-add/subtract based beamformer, the direction θ of maximum/minimum amplification is varied, i.e. the reception lobe of the beamformer is angularly shifted. As also disclosed in the WO 00/68703 and also preferably applied to the overall of the present invention, signal processing is performed in frequency mode and frequency-specifically. At the output of the histogram-forming units 44 the instantaneously prevailing DOA-dependent histograms are present and signals or data dependent there from are fed to a histogram classification unit 46. Therein, the histogram courses resulting from left ear and right ear acoustical signal reception are evaluated, thereby preferably including comparing the histogram courses as prevailing at the units 44_L, 44_R.

[0061] In unit 46 on one hand the histogram courses per se are evaluated, e.g. and with an eye on fig. 8 on peaks, width of the peaks, time behavior of the peaks etc., and the acoustical surrounding with respect to acoustical sources therein is respectively classified, as e.g. under the aspect of "acoustical source moving away", "acoustical source moving in the surrounding", "acoustical source becoming less relevant", "new acoustical source appearing", "acoustical source disappearing", "head of the individual moving", etc. Additionally the interrelation of both histogram courses is evaluated, thereby detecting how one of the histogram courses alters or appears with respect to the other side histogram course. This is for instance caused by the re-

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spective $HRTF_L$ and $HRTF_R$ becoming at the left and right ears (L, R) differently effective in dependency of DOAφ. Instead of performing classification on the basis of DOA according to the second aspect of the present invention other classifications may be exploited as for instance described in the WO 02/32208 of the same applicant which accords with the US application no 10/059 059.

[0062] At the output of histogram classifying unit 46 there are generated control signals or data dependent on the classification result and from preset classification-dependent settings to be realized at the hearing device system. Thereby at the output of classification unit 46 a signal or data is generated, which is operationally connected to the beamformer control input BFC_L and BFC_R and on the other hand there is generated a control signal or data input to the weighting unit 49, which accords to the unit 9_w of the system of fig. 7. The beamformer control data and respective output is shown at BFC in fig. 9, the weighting unit control signals or data and respective output of unit 46 by SC. The SC signals or data do control, as was more generically shown in fig. 6 at the output of unit 12_w, the weighting unit 49 in that, shown by varying weighting coefficients α to γ in Fig. 6, the weights or transfer functions with which the output signals at outputs A_{1L}, A_{1R} respectively act upon electrical/mechanical converters 47_L and 47_R.

[0063] To further explain the embodiment of fig. 9 let us make an example. To start with there shall appear in the $\varphi = 0$ DOA-direction with respect to the units 40 a significant acoustical source. The beamformers of the units 40 have their lobe directed on that source defining for $\varphi = \theta = 0$. Both histograms at unit 44 may have e.g. a course as shown in fig. 8(a). The histogram classification unit 46 recognizes histogram peaks for $\varphi = 0$ at both histograms, and this defines at unit 46 for a yet stable and significant acoustical source. Accordingly by means of BFC the beamformers are kept on $\theta = 0$. The SC control signal controls the selection unit 49 for equally weighted influence of signals or data appearing at both outputs A_{1L} A_{1R} upon the converters 47.

[0064] Now let's assume this relevant acoustic source in the acoustical surrounding U starts to move to the right-hand side of fig. 9. This is recognizable at unit 46, because both histogram courses will show a development according to fig. 8(d). Thus, unit 46 recognizes: "source is moving to the right". As the acoustical source considered leads still to a significant sharp peak in both histogram courses, the beamformers of units 40 are both controlled by the control signals or data BFC to follow that source. Still the SC control signals control selection unit 46 at least nearly for equally distributed weighting of the influence of the output signals A_{1L} and A_{1R} upon the converters 47_L and 47_R.

[0065] As the acoustical source moves further to the right the head-related transfer function HRTF starts to influence the acoustical signals as impinging on the units 40. Whereas the right-hand side received acoustical signals will not be affected by the HRTF, the left-hand side received acoustical signals from that source become more and more influenced by HRTF as the acoustical source becomes "hidden" by the individual's head H. Therefore, the histogram course at unit 44_R will still have a pronounced peak representing the source considered, whereas due to the HRTF the histogram course at unit 44_L will show at the angular position of the source considered, which is equal to the angular position of the peak in the histogram course at unit 44_R, a more and more enlarged, less pronounced peak. This is, purely as an example, shown in fig. 9 aside the histogram-forming units 44 and with respect to the same angular position φ_s of the acoustical source considered. The classifying unit 46 recognizes by comparing the two histogram courses that at the same angular position φ_s the left side histogram course has a widened and less pronounced peak with respect to the right-hand histogram course. This indicates the type of acoustical surrounding according to which a moving acoustical source has moved so far to the right that the respective HRTF function becomes effective. This means that the data from that source processed in the left ear unit 40_L become less accurate than the data processed in the right ear unit 40_R from that source and therefore the selection unit 49 is controlled to react on this specific exemplified situation by increasing the influencing of the right side signals or data at output A_{1R} upon the converters 47_L and 47_R. Thereby and e.g. within unit 49 the $HRTF_L$ function, which takes effect on the acoustical signals impinging upon the left side unit 40_L, will be maintained with respect to data operationally acting upon converter 47_L in a most preferred mode, so as to maintain for the individual spatial perception of the acoustical source. With respect to beam control, as the DOA data of the right ear unit 40R become according to this example more accurate than the respective data from unit 40_L e. g. due to higher level acoustic signals, also beamformer control will preferably be at least dominated by the DOA data from the right ear unit 40_R (not specifically shown in fig. 9).

[0066] The weighting-coefficients or functions as of α to γ of fig. 6, are preferably complex valued, frequency or frequency band dependent functions. In the classifier unit also multiple acoustical source situations are detected and predetermined strategies are set, how to control on one hand the beamformers, on the other hand the signal transmission at weighting unit most suitably for specific acoustical surroundings.

[0067] Thus, by combining the two aspects of the present invention a binaural hearing device system is achieved, which incorporates "intelligent" system adjustment based on the evaluation of DOA histogram course.

[0068] Once again it must be emphasized that the data or signal processing functions which have been explained as by fig. 9 may be split in a great variety of realization modes to the two hearing devices or may be

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centralized within a unit remote from the hearing devices, and accordingly the signal transmission link 5 from one ear side to the other will be provided. Further, the skilled artisan recognizes that the system as of fig. 9 will incorporate different digital processing unit DSPs, especially along the double-arrowed operational connections so as to take into account specific hearing improvement needs at both individual's ears, HRTF functions etc.

[0069] As we have mentioned before one approach, which is today a preferred one, for and as a second aspect of the present invention is to provide classification of the acoustical surrounding of an individual so as to appropriately control a hearing device, being it a monaural or a binaural hearing device, based on evaluation of the direction of arrival DOA.

[0070] An approach how to determine the DOA is, as was explained before, explained in detail in the WO 00/68703. Based on that teaching, in fig. 10 there is exemplified a binaural hearing device system whereat on one hand and according to the first aspect of the present invention combined data or signals from at least two input acoustical/electrical converters are respectively transmitting from one ear side to the other or in the case of a CIC-device with one input converter after having been processed by a Wiener-Filter. On the other hand the embodiment of fig. 10 incorporates also the second aspect of the present invention realised on the basis as disclosed in the WO 00/68703. A left ear reception unit 50_L comprises two beamformers one defining a maximum amplification characteristic in DOA = 0° direction, the other one in the backwards DOA = 180° direction. In fig. 10 the beamformers are exemplified as being equal first order cardoid beamformers.

[0071] Unit 50_L outputs at respective outputs A_{50L1} and A_{50L2} signals or data dependent on the impinging acoustical signals amplified by the respective DOA dependent amplification of the beamformers and frequency dependent.

[0072] These signals are respectively denoted in fig. 10 by S_{F1} and S_{B1}. This output signals are led after analogue/digital conversion (not shown) to time domain/frequency domain conversion units 52_{L1} and 52_{L2} resulting in frequency specific output signals or data C_{B1} and C_{F1}. Signals dependent from the output signals of the conversion units 52 are further fed to absolute value forming units 54_{L2} and 54_{L1} outputing respective frequency specific signals or data $|C_{B1}|$ and $|C_{F1}|$. These absolute value signals or signals dependent there from are fed to a quotient forming or division unit 56_L outputing for left ear reception unit 50_L frequency specific a quotient Q_L. Signals or data dependent from that quotient Q_L are subjected to histogram forming in a histogram forming unit 58_L outputing of histogram data H_L.

[0073] The right ear side with right ear reception unit 50_R up to data H_R is preferably construed exactly equally to the left ear side as just described and will therefore not specifically be described again.

[0074] The histogram data from the two histogram forming units 58_L and 58_R are input to a classifying unit 60.

[0075] Further, signals dependent on the front-forwards beamformers at both reception units 50_L and 50_R namely $|C_{F1}|$ and $|C_{F2}|$ are fed to a further quotient forming unit 62_v and in analogy signals dependent from the output signal of the rear beamformers of both reception units as of $|C_{B1}|$ and $|C_{B2}|$ are fed to still further quotient forming unit $62R_e$. Signals or data dependent from the result at the said quotient forming units 62_v and 62_{Re} are input to respective histogram forming units 64_{Re} and 64_v. The histogram data output by these histogram forming units are again input to the classification unit 60.

[0076] After classification, e.g. as will just be discussed, the classification unit 60 generates output signals or data which are operationally linked to a control input of the weighting unit 61. As a function of the classification result-data output by classification unit 60 signal transfer within weighting unit 61 is controlled, namely:

- from an input E_{L1} to which signals dependent from the forward beamformer of unit 50_L are fed to output A_L and output A_R respectively,
- from an input E_{L2} to which signals or data dependent from the output signals of the rear beamformer of unit 50_L are fed respectively to the output A_L and A_R
- and in complete analogy, from the right ear input E_{R1}, E_{R2} and to the said respective outputs A_L and A_R. The signals output at A_L and A_R are operationally fed to the output electrical/mechanical converters 63_L and 63_R respectively.

[0077] We define:

$$Q_L = \frac{|C_{F1}|}{|C_{B1}|}$$

$$Q_R = \frac{|C_{F2}|}{|C_{B2}|}$$

$$Q_{Re} = \frac{|C_{B1}|}{|C_{B2}|}$$

$$Q_V = \frac{|C_{F1}|}{|C_{F2}|}$$

[0078] Let's discuss possible classification results and criteria exploited and generated at unit 60 whenever an acoustical signal source in the surrounding U is detected with different DOA's.

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[0079] Whenever DOA φ is between 0° and 90° following is valid:

$$Q_L > 1 \text{ and } Q_V > 1.$$

[0080] It has to be noted that it is preferred to consider Q_V in this case than Q_{Re} because the acoustical signal impinges at the higher level on the forward beamformer of both units 50, the output signals of these beamformers being thus more accurate with respect to signal/noise than the output signals of the respective rear side beamformers.

[0081] The same is considered with respect to evaluating Q_L or Q_R, the signals leading to Q_L have a better signal/noise ratio than the signals leading to Q_R because as the target acoustic source moves towards 90° the right side HRTF more and more influences signals received at the right ear unit 50_R. These considerations are made also in the following cases to be discussed and are not repeated.

[0082] As the target source is located at the DOA φ between 90° and 180° the following is valid:

$$Q_L < 1 \text{ and } Q_{Re} > 1.$$

[0083] As the target source moves on to a DOA φ between 180° and 270° the following prevails:

$$Q_R < 1 \text{ and } Q_{Re} < 1.$$

[0084] Finally as the target source moves to a position between 270° and 360° the following prevails:

$$Q_R > 1 \text{ and } Q_V < 1.$$

[0085] Thus by evaluating these criteria, as a simplified example, within the classification unit 60 it is established around 360° where an acoustical source is located and accordingly in weighting unit 61 the respective signal transfer functions are set. As an example:

[0086] If the source is detected by the above criteria to be located at a DOA between 90° and 180° the rear side beamformer of left ear reception unit 50_L will become master beamformer because that beamformer outputs a signal with best signal/noise ratio. Therefore the transfer functions or coefficients according to fig. 6 from input E_{L2} on the one hand to A_L and on the other hand to A_R will become governing. Thereby the transferred function from E_{L2} to A_R will consider the HRTF which is not influencing at the source position discussed signals impinging on the reception unit 50L but which must be considered for driving the right output converter 63R so as to maintain spatial source perception. Simplified the forward beamformer of unit 50L and both beamformers at unit 50R become slaves and their respective output signals are merely exploited to generate the respective quotients to allow the classification unit 60 to properly classify the prevailing DOA so as to properly control signal transfer in weighting unit 61.

Claims

1. A binaural hearing device system comprising a first device for one ear of an individual, a second device for the other ear, a data communication link between said first and said second devices, said first device comprising at least a reception unit with at least two input acoustical/electrical converters and a signal processing unit, the inputs thereof being operationally connected to the electrical outputs of said at least two converters and generating at an output a signal dependent on signals at both said inputs, said communication link being provided at the output side of said processing unit and transmitting signals dependent upon said output signal of said processing unit said second device comprising at least a output electrical/mechanical converter.

2. The binaural hearing device of claim 1, wherein said first device is a device to be completely introduced into individual's ear channel (CIC), wherein instead of said at least two input converters there is provided a single acoustical/electrical input converter and wherein instead of said processing unit with at least two inputs there is provided a processing unit with one input operationally connected to the output of said single input converter, said signal processing unit performing at least a Wiener filter operation upon the signal applied to said input.

3. The system of one of claims 1 or 2, wherein said first device for said one ear does not comprise an electrical/mechanical output converter.

4. The system of claim 1 or 2, wherein said second device for said other ear does not comprise an input acoustical/electrical converter.

5. The system of claim 1 or 2, wherein said first device for said one ear comprises an output electrical/mechanical converter unit, the input thereof being operationally connected to the output of said processing unit.

6. The system of claim 1 or 2, wherein said reception unit is a first reception unit, said at least two input acoustical/electrical converters are first acoustical/electrical converters at a first reception unit, said signal processing unit is a first signal processing unit, said output electrical/mechanical converter is a second output electrical/mechanical converter,

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said first device comprising a first output electrical/mechanical converter, said second device comprising a second reception unit with at least one second input acoustical/electrical converter.

7. The system of one of claims 1 or 2, wherein said data communication link is a wire-bound, an optical fiber or a wireless communication link.

8. The system of claim 6, wherein said second reception unit comprises at least two second input acoustical/electrical converters and a second signal processing unit.

9. The system of claim 8, wherein the inputs of said second signal processing unit are operationally connected to the outputs of said second input converters and generates at a second output a signal dependent on signals at both said inputs of said second signal processing unit, said data communication link being provided additionally at the output side of said second signal processing unit.

10. The system of claim 8 or 9, the output of said first signal processing unit being operationally connected to a first input of a weighting unit, the output of said second signal processing unit being operationally connected to a second input of said weighting unit, said weighting unit having a first output operationally connected to the input of said first output converter and a second output operationally connected to the input of said second output converter, said weighting unit having a control input, said weighting unit varying operational connection of said first input to said first output, from said first input to said second output, from said second input to said first output and from said second input to said second output, controlled by a signal applied to said control input.

11. The system of claim 10, wherein said operational connections comprise frequency dependent, complex transfer functions.

12. The system of claim 10, wherein said control input is operationally connected to the output of a classification unit with at least one input operationally connected to at least one output of at least one of said reception units.

13. The system of claim 1, wherein said first device comprises a beamformer unit with a beamcontrol input and with an output, a detection unit for the direction of arrival of an acoustical signal impinging upon said reception unit and generating an output signal in dependency of said direction of arrival at an output, said output of said direction of arrival detection unit being operationally connected to said beamcontrol input of said beamformer unit.

14. The system of claim 12, further comprising a determination unit for the direction of arrival of an acoustical signal said determination unit being interconnected between said at least one input of said classification unit and said at least one output of said at least one reception unit.

15. The system of claim 14, further comprising at least one histogram forming unit the input thereof being operationally connected to said at least one output of said at least one reception unit, the output thereof being operationally connected to an input of said classification unit.

16. A method for controlling a hearing device system, comprising at least a reception unit at a first device for one ear having at least two input acoustical/electrical converters and at least an output electrical/mechanical converter at a second device for the other ear and a communication link between said first and said second devices, comprising the steps of generating in dependency of output signals of said at least two input converters a combined signal and transmitting said combined signal via said communication link.

17. The method of claim 16, further comprising the step of providing instead of said at least two input converters only one converter and construing said first device as a device to be completely introduced into the ear channel, further comprising to step of treating the output of said one input converter by a Wiener-Filter and transmitting signals dependent from the output of said Wiener-Filter via said communication link.

18. The method of claim 16 or 17, further comprising the step of not providing an electrical/mechanical output converter at said first device.

19. The method of claim 16 or 17, further comprising the step of not providing an input acoustical/electrical converter at said second device.

20. The method of claim 16 or 17, further comprising the step of providing an output electrical/mechanical converter to said first device.

21. The method of claim 16 or 17, further comprising the step of providing at least one input acoustical/electrical converter at said second device and operating said output electrical/mechanical converter at said second device in dependency of said signal transmitted via said communication link and an output signal of said at least one input acoustical/electrical converter of said second device.

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22. The method of claim 21, further comprising controlling signal transfer functions to a signal driving said output electrical/mechanical converter at said second device on one hand from output signals of said at least two input converters of said first device and on the other hand from an output signal of said at least one input converter of said second device.

23. The method of claim 22, further comprising the step of classifying signals dependent from at least two of the output signals of said at least two input converters at said first device and of said at least one input converter at said second device and controlling said transfer functions in dependency of a result of said classifying.

24. The method of claim 23, further comprising the step of determining direction of arrival of an acoustical signal impinging on said devices and performing said classification in dependency of said direction of arrival.

25. The method of claim 24, further comprising the step of generating at least one histogram indicative of time occurrence of directions of arrival and performing said classification in dependency of said histogram.

26. The method of claim 21, further comprising the step of providing at said first device an output electrical/mechanical converter.

27. The method of claim 26, further comprising the step of driving said output converter of said second device in dependency of output signals of said at least one of at least two input converters of said first device and in dependency of output signals of said at least one input converter of said second device and driving said output converter at said first device in dependency of output signals of at least one of said at least two input converters at said first device and in dependency of output signals of said at least one input converter of said second device.

28. The method of claim 27, further comprising the step of controlling said signal transfer functions of said dependencies between output signals of said input converters and said output converters.

29. The method of claim 28, further comprising performing a classification of signals dependent on at least two output signals of said at least two input converters of said first device and said at least one input converter of said second device and controlling said dependencies between said output signals of said input converters and said output converters in dependency of results of said classification.

30. The method of claim 29, further comprising determining direction of arrival of acoustical signals upon said devices, said classification comprising classifying of said direction of arrival.

31. The method of claim 29, further comprising the step of forming at least one histogram of a signal, said classification comprising classifying the result of said histogram forming.

32. The method of claim 26, said second device comprising at least two input converters.

33. A method for producing a drive signal for an electrical/mechanical output converter of a binaural hearing device comprising the steps of acoustical/electrical converting impinging acoustical signals at at least two input converters of a device to be applied adjacent or in individual's one ear, transmitting a signal dependent from both said convertings via a link to a further device to be applied adjacent or in individual's other ear and generating said drive signal in dependency of said transmitted signal.

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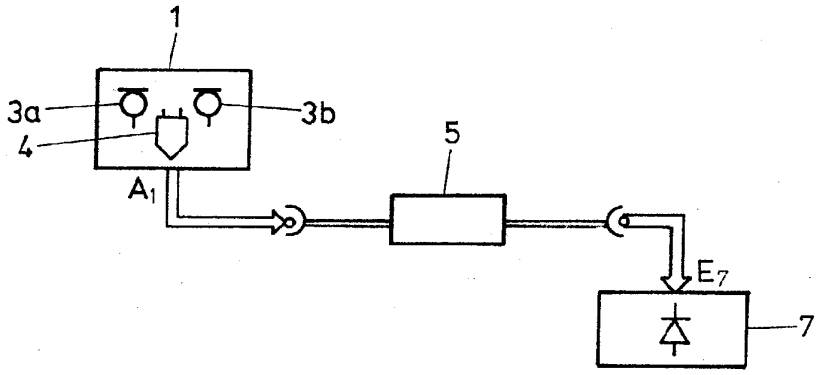


FIG.1

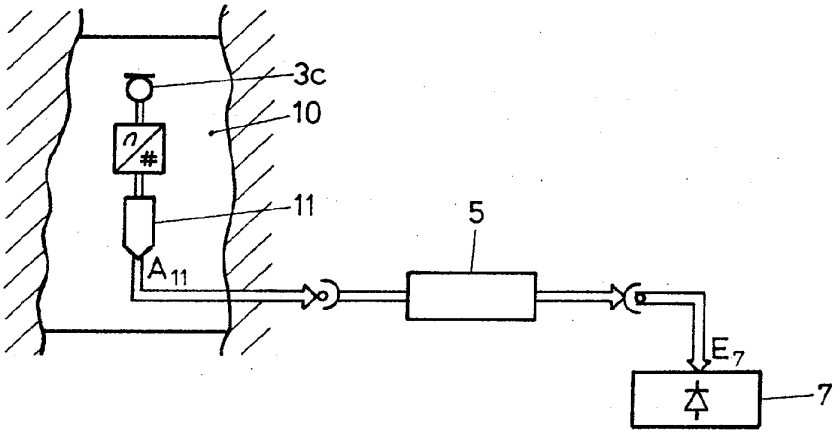


FIG. 2

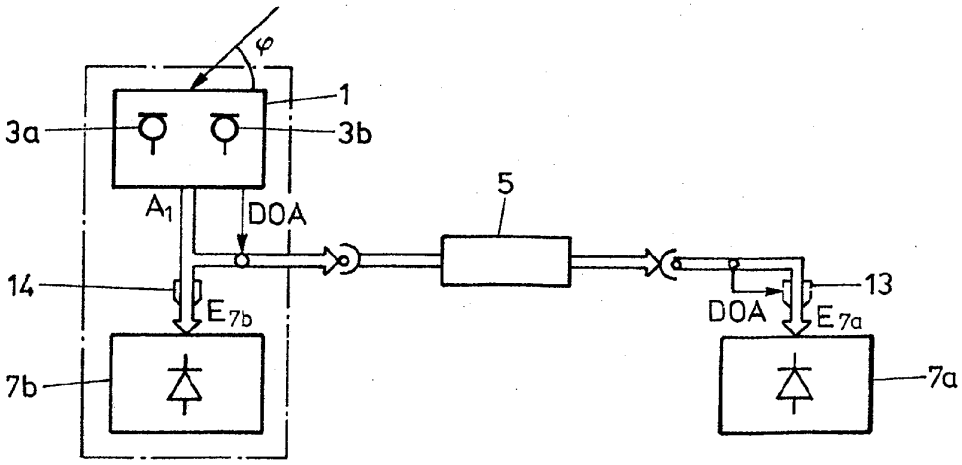


FIG.3

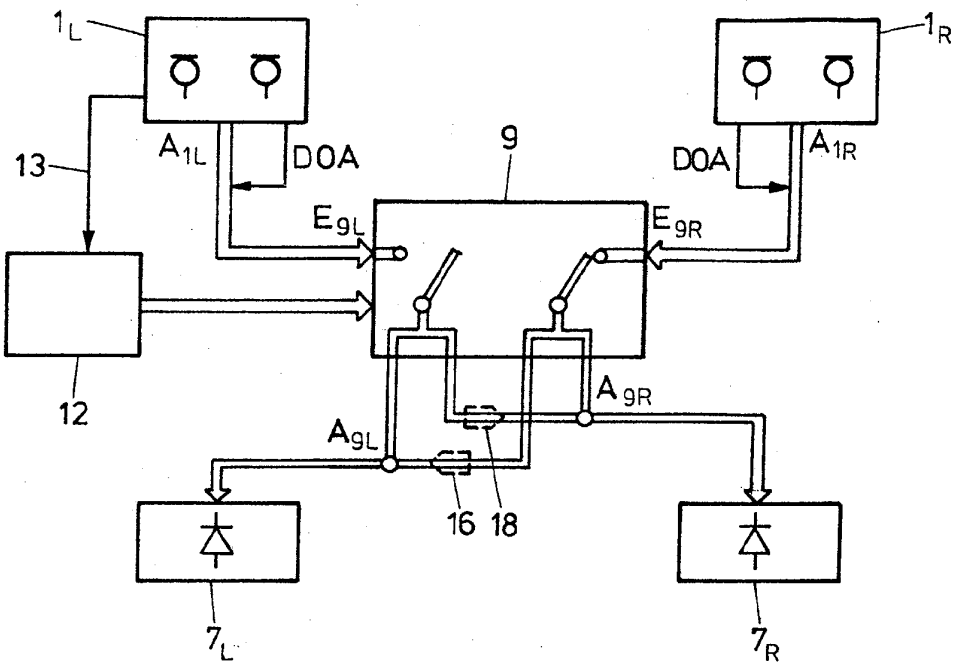


FIG. 4

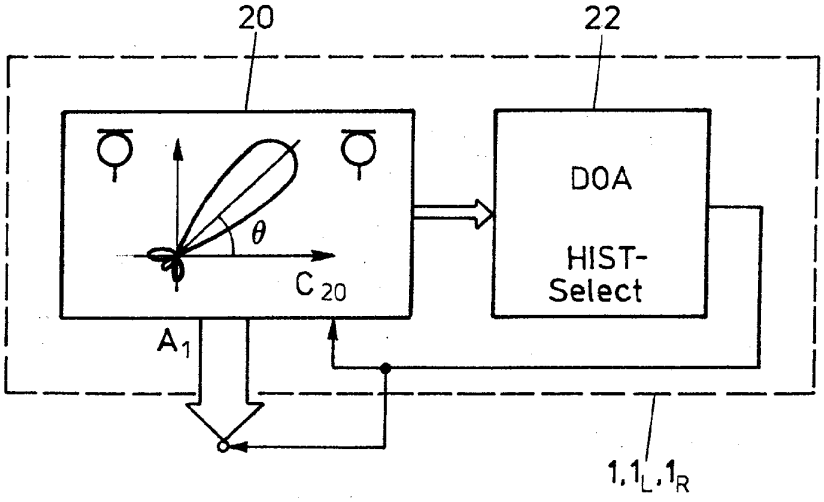


FIG. 5

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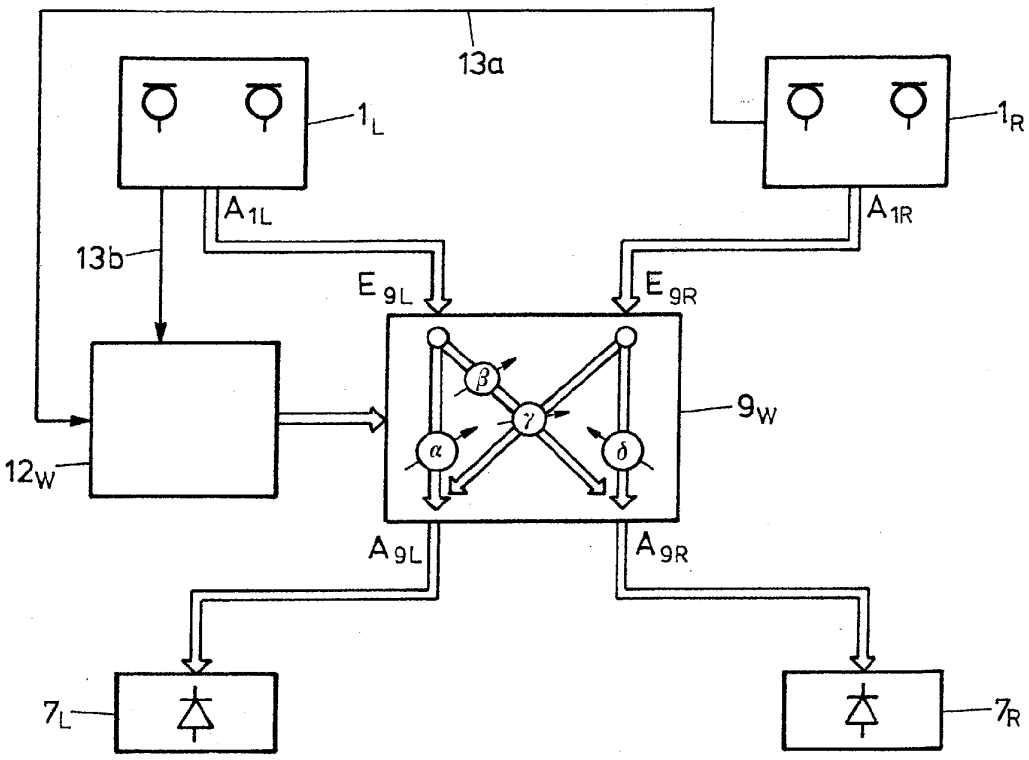


FIG.6

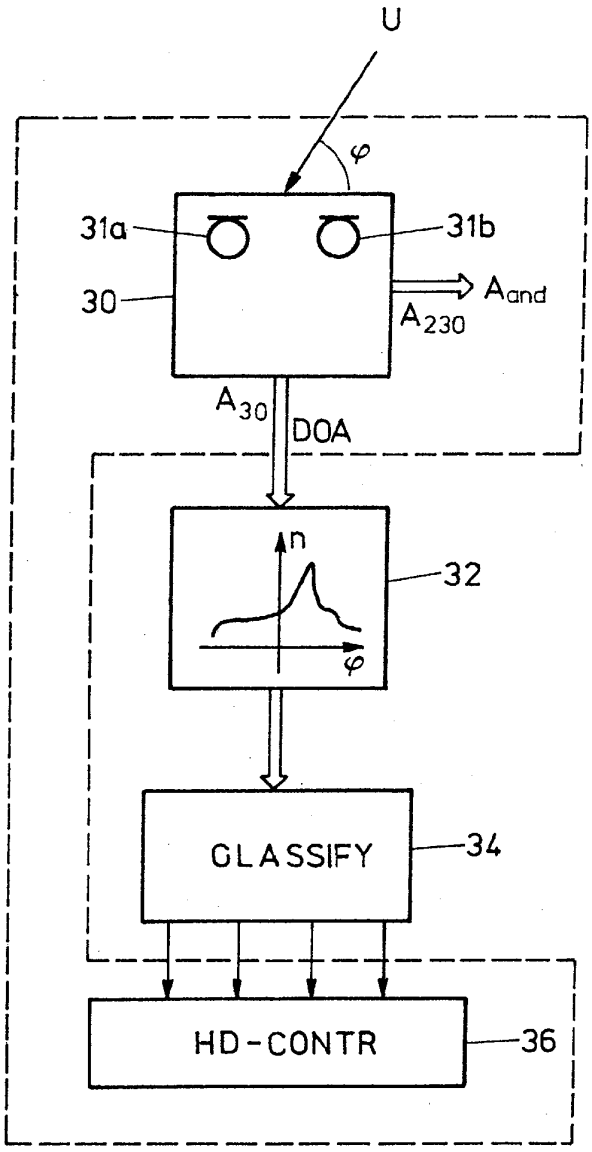


FIG.7

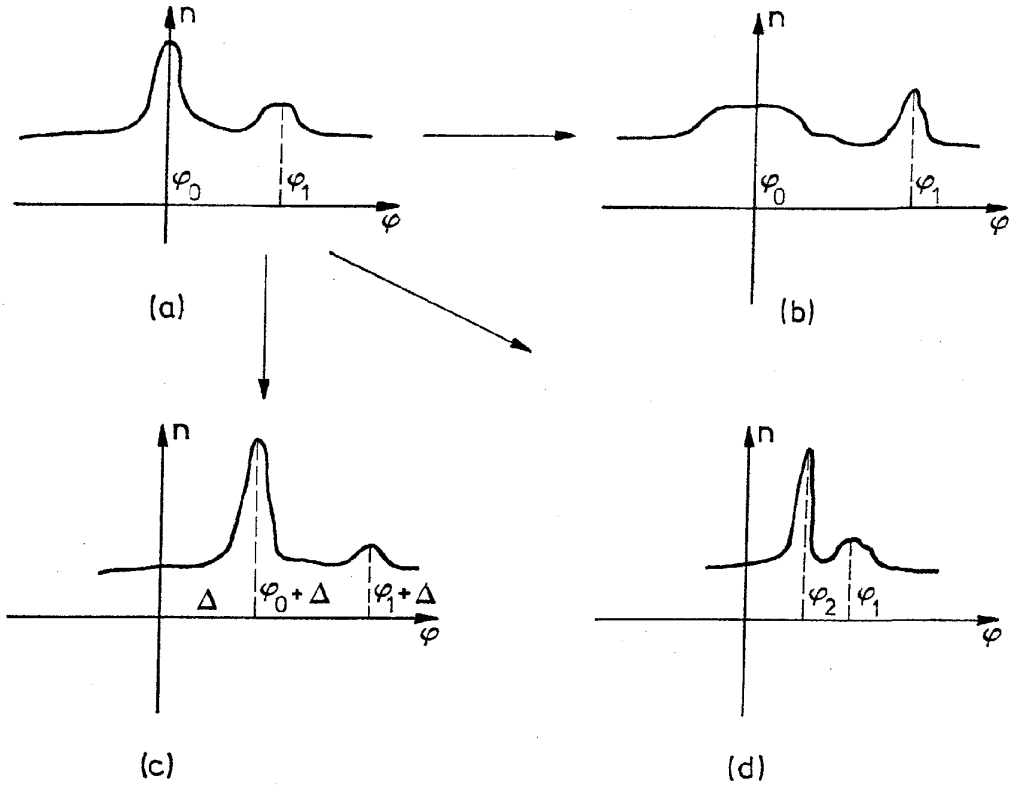


FIG.8

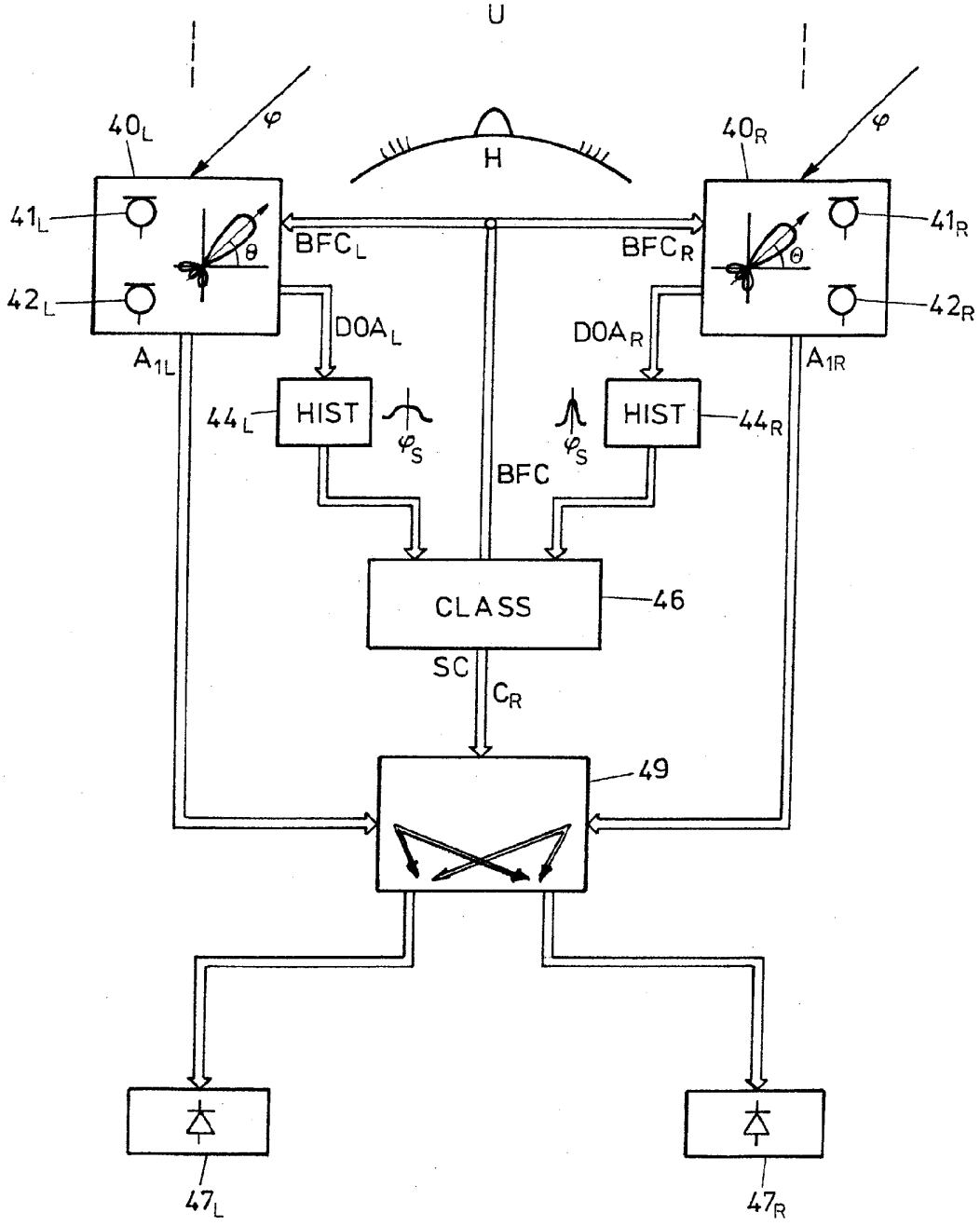

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FIG.9

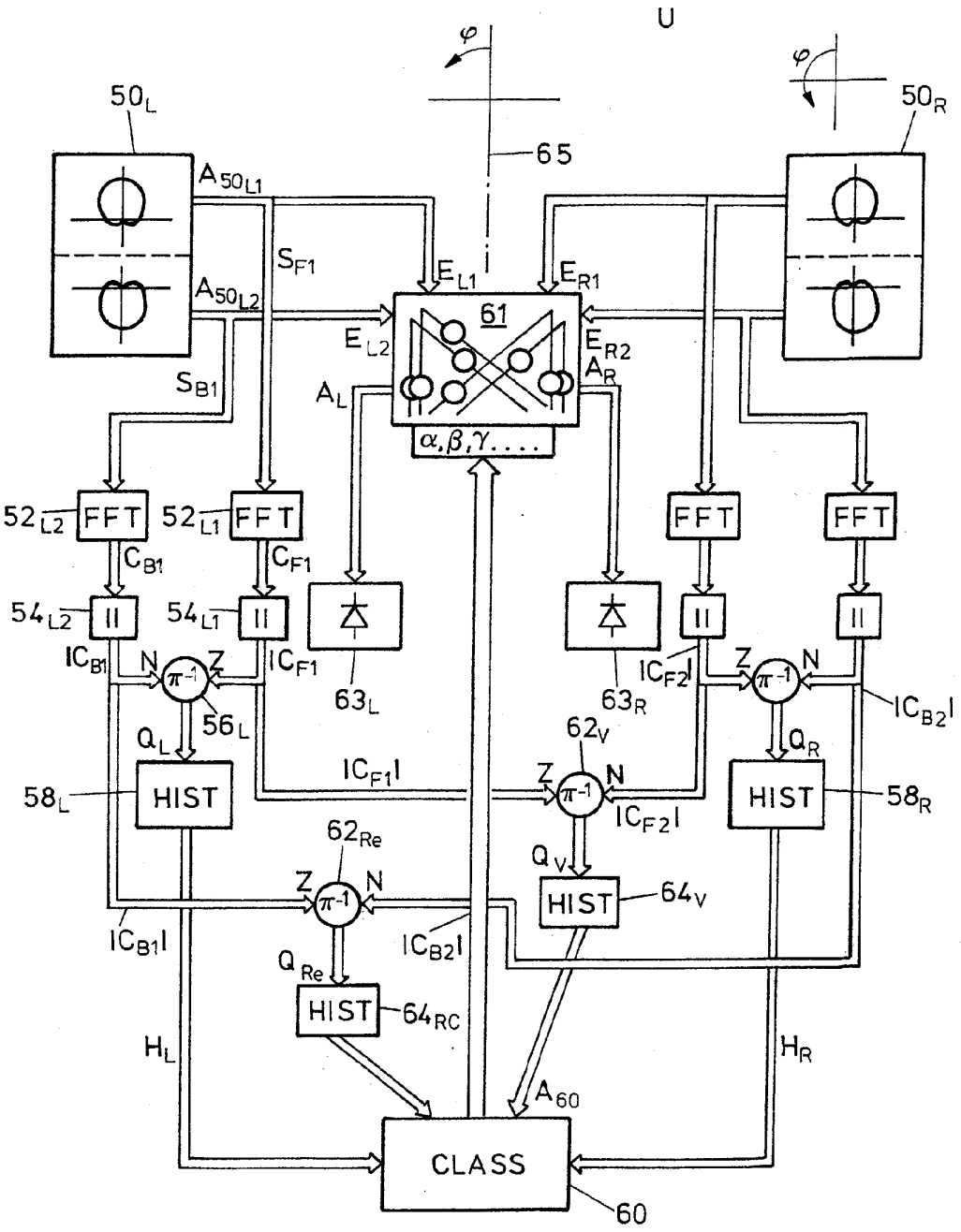


FIG.10

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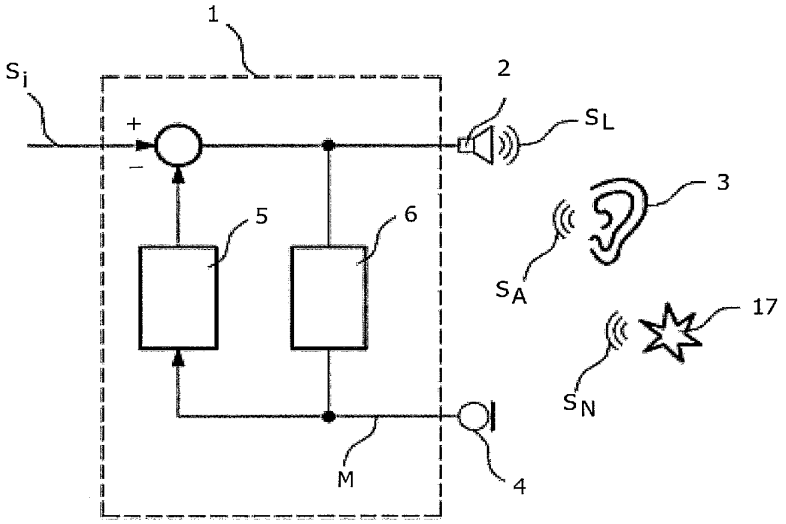
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— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))*

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(54) Title: AUDIO DEVICE FOR AND METHOD OF DETERMINING BIOMETRIC CHARACTERISTINCS OF A USER.



(57) Abstract: The invention describes an audio device (1), comprising a loudspeaker (2) for outputting a loudspeaker output signal (S_L) close to the ear (3) of a user of the audio device (1), a first microphone (4) arranged in a manner suitable for recording a first microphone input signal (M) including an aurally reflected signal component (S_A) and an active noise cancellation circuit (5) for actively reducing any background noise component (S_N) of the microphone input signal (M) to give a noise-reduced microphone input signal (M). Aural biometric characteristics of the user are determined by a biometric characteristics determination unit (6) on the basis of the noise-reduced microphone input signal (M).

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Audio device for and method of determining biometric characteristics of a user

FIELD OF THE INVENTION

The invention relates to an audio device for determining biometric characteristics of a user.

The invention further relates to a method of determining biometric characteristics of a user.

The invention further relates to a system for biometric identification.

The invention further relates to a method for identification of a user on the basis of aural biometric characteristics of that user.

BACKGROUND OF THE INVENTION

Usually, the term "biometrics" is used to describe methods of recognizing a person based on physiological or behavioural characteristic of that person. Such a characteristic is referred to as a biometric, and can be the person's face, his fingerprints, hand geometry, handwriting, iris, retinal pattern, vein pattern, voice, etc. The use of human biometric features for identity verification and person recognition is well known. Biometric technologies are becoming the foundation of an extensive array of highly secure identification and personal verification solutions. The advantage of using biometrics for access control, rather than using other methods such as PIN (personal identification number) codes or identity cards, is that a biometric cannot be lost, given away, or stolen from the person it characterizes. However, not all biometric features are immune to being copied. For example, it is possible to make a copy of a fingerprint and to fraudulently use this copy to gain access to data otherwise only accessible to the true owner of the fingerprint. Furthermore, some types of biometric identification systems fail to gain general acceptance by the public. For example, a person may feel uncomfortable having to expose his eye to a retina or iris scanner.

It is known from prior art that the acoustical properties of the ear can be used to identify people uniquely. This kind of biometric feature cannot easily be copied, and can easily be implemented in a mobile phone for remote identification, thus replacing conventional, less reliable methods of identification such as the PIN code. In the case of acoustic ear canal biometrics, what is of interest is the topology of the ear canal, which is unique for every human. An incoming sound signal is reflected and otherwise modified by the ear canal to give an aurally reflected signal which exits the ear canal. In US 5,787,187 a

sound signal is directed into the ear of a user, and the frequency response of the ear canal is measured and analysed to extract a feature vector unique to this user. However, since the microphone used to detect the response from the ear canal must also pick up any surrounding sound signals, such a measurement system is particularly prone to error owing to background noise. These unwanted background noise signals can really only be excluded from the measurement described by, for example, enclosing the microphone and the ear in headphones of a size large enough to encompass the entire ear. Since such headphones are generally cumbersome to use and awkward to transport, they are impractical for frequent use, and unsuited to user identification for applications such as telephone banking, telephone brokerage, etc, which a user generally wishes to carry out with a mobile phone, whether at home or underway.

SUMMARY OF THE INVENTION

Therefore, an object of the present invention is to provide a robust and user-friendly way of identifying a user on the basis of his aural biometric characteristics.

To this end, the present invention provides an audio device, comprising a loudspeaker for outputting a loudspeaker output signal close to the ear of a user of the audio device and a first microphone arranged in a manner suitable for recording a first microphone input signal including an aurally reflected signal component, an active noise cancellation circuit for actively reducing any background noise component of the microphone input signal, and a biometric characteristics determination unit for determining the aural biometric characteristics of the user on the basis of the noise-reduced microphone input signal.

An appropriate method for determining biometric characteristics of a user according to the present invention comprises directing a loudspeaker signal into the ear of a user and recording the response as a microphone input signal. The microphone input signal will include other signal contributions besides the loudspeaker signal, such as an aurally reflected signal component contributed by reflections of the loudspeaker signal back out of the ear canal of the user. If the microphone also picks up any other sounds generated in the background – which detract from the quality of the microphone signal and are therefore collectively referred to as background noise – then, according to the invention, such a background noise signal component of the microphone input signal will be reduced by an active noise cancellation circuit. The aural biometric characteristics of the user are then determined using the resulting noise-reduced microphone input signal.

Using the invention, the audio device can be any type of audio device with a

loudspeaker which is intended for use close to the ear of the user, such as a telephone handset, a mobile telephone, a set of headphones, etc., and does not require special headphones to exclude background noise. The loudspeaker incorporated in or on the audio device can issue sound signals essentially directly into the ear canal of the user. Since the first microphone is intended to pick up the loudspeaker signal reflected back out of the ear canal – the aurally reflected signal – the microphone need only be positioned so that it is advantageously placed for picking up these reflected sound signals. Preferably, the microphone might be placed in the direct vicinity of the loudspeaker, but it may even be positioned within the circumference of the loudspeaker itself.

Furthermore, using the invention, it is not necessary to use the audio device under ideal conditions. Any unwanted signal components of the microphone input signal – such as traffic noise, conversations being carried out in the vicinity, music etc. – detracting from the quality of the microphone input signal, and collectively referred to as a background noise signal, can essentially be "cancelled" by isolating the background noise signal, phase-inverting it and subtracting it from the loudspeaker output signal to give a noise-reduced microphone input signal. This technique is known as active noise cancellation (ANC). ANC circuits are already to be found in a variety of audio devices, e.g. mobile phones, where they are used to optimise the quality of the loudspeaker output signal, allowing the audio device to be used in a noisy environment, since the background noise is suppressed at the position of the user's ear.

According to the invention, this technique is used to yield a noise-reduced microphone input signal, now comprising essentially the loudspeaker output signal and the aurally reflected signal component, which is forwarded to the biometric characteristics determination unit. Here, with the aid of suitable filters, biometric characteristics of the user can be determined on the basis of the aurally reflected signal component of the microphone input signal, as will be explained in more detail below.

A particularly advantageous characteristic of the present invention is, therefore, that an audio device comprising the necessary loudspeaker, microphone, ANC circuit and filters can easily be also used for biometric user identification without requiring extensive additional hardware. For example, a mobile phone which incorporates an additional microphone in the vicinity of the loudspeaker might be used, whereby it is not necessary to ensure particularly favourable environmental conditions, since the presence of background noise becomes essentially irrelevant when the invention is applied.

The dependent claims and the subsequent description disclose particularly

advantageous embodiments and features of the invention.

A sound signal can comprise a number of frequency components, and can be described in the frequency domain by its frequency spectrum which shows to what extent each frequency is represented in the sound signal. When an acoustic signal is reflected or modified in some way by the surrounding environment, the resulting signal is called the response signal. In travelling from a source (loudspeaker) to a target (microphone) in a non-ideal environment, a sound signal suffers some degree of attenuation. Furthermore, the quality of the loudspeaker and microphone also influence the quality of the received signal. Some frequencies, typically the higher frequencies, will generally suffer more attenuation than others. The frequency response for a recorded response signal can be calculated by performing a Fourier transform on the recorded response signal. Typically, this is carried out digitally on a sampled response signal using a digital filter to perform a discrete Fourier transform, which can be further developed to derive a "feature vector" characterizing the source, target and surrounding environment. Attenuation effects of the microphone are also included in this feature vector. The net changes which the original signal undergoes from source to target can be expressed, generally in the frequency domain, using a "transfer function", which essentially describes the quality of signal transfer from source to target. For example, for a loudspeaker/microphone pair, the transfer function is termed the "loudspeaker to microphone transfer function", and takes into account the frequency response of the microphone, as well as the frequency response of the loudspeaker.

In the case of the ear canal, a sound signal directed into the ear canal will be reflected back out in an attenuated form. The amount of attenuation is unique to each ear canal, so that the feature vector of the frequency response of the ear canal can be used as a reliable biometric characteristic for that ear canal (and ultimately, for the user), and the transfer function from the loudspeaker to the ear is given by the loudspeaker to ear transfer function. Since the noise-reduced microphone input signal essentially consists of the loudspeaker output signal and the aurally reflected signal component, by processing the noise-reduced microphone input signal in a suitable manner it is possible to obtain the loudspeaker to ear canal transfer function, which comprises the biometric characteristics for this user. Regardless of the quality of the loudspeaker output signal, the feature vector calculated from the frequency response of the ear canal will always be the same for a particular user.

In a preferred embodiment of the invention, an adaptive filter arrangement might be used to process the noise-reduced microphone input signal and to determine the biometric characteristics for the user. The adaptive filter arrangement might comprise an analog or

digital filter for processing the loudspeaker signal, and an update part for modifying the filter, for example by modifying the coefficients of an adaptive digital filter or the inductive, capacitive, or resistive parts of an adaptive analog filter. The input signal to the update part might be obtained by subtracting the filtered loudspeaker output signal from the noise-reduced microphone input signal. An adaptive filter arrangement such as the type often used for acoustic echo cancellation (AEC) might be used. Therefore, the audio device can preferably make use of existing circuitry, otherwise used to perform acoustic echo cancellation in the audio device, to also calculate the biometric characteristics of the user's ear canal.

In a particularly preferred embodiment of the invention, the adaptive filter arrangement – or biometric characteristics determination unit – of the audio device might be used to indirectly improve the quality of the microphone input signal by modifying the behaviour of the active noise cancellation circuit. Therefore, a control signal connection is preferably implemented between the biometric characteristics determination unit and the active noise cancellation circuit, such that a control signal generated by the biometric characteristics determination unit controls the behaviour of the active noise cancellation circuit. In this way, the noise reduction or cancellation function of the active noise cancellation circuit can be improved to give a better noise-reduced microphone input signal, which in turn provides for a more accurate estimate of the feature vector.

In a preferred embodiment of the invention, the audio device might comprise a second microphone arranged in a manner suitable for picking up or recording a second microphone input signal component, such as a user speech signal, for transmission to a conversation partner. The audio device might therefore be used in the manner of a telephone handset, a mobile phone, a headset, etc. The user of such an audio device can listen to and speak to his conversation partner using the loudspeaker and second microphone respectively.

Since the second microphone might pick up the loudspeaker output signal as well as the desired speech signal component, a preferred embodiment of the invention provides for an acoustic echo cancellation arrangement for reducing any loudspeaker output signal component in the second microphone input signal. This improves the listening quality for the users of such audio devices.

Since the function of the biometric characteristics determination unit may effectively be the same as that of an acoustic echo cancellation circuit, it might be sufficient to implement just one such circuit in the audio device. The single acoustic echo cancellation circuit can be used to minimize the loudspeaker contribution to the second microphone input

signal while one of the conversation partners is talking, and it can also be used to determine the aural biometric characteristics of the user, for example, during periods of silence. Therefore, in a particularly preferred embodiment of the invention, a switch is implemented for switching an acoustic echo cancellation arrangement between the first microphone input signal and the second microphone input signal. Switching between microphones can be effected once, for example when the user of the audio device initially establishes a connection between that audio device and a remote device, or at intervals, for example during periods of silence. In this way, even the speech signal of the conversation partner can be used in determination of the aural biometric characteristics of the user. The aural biometric characteristics of the user are preferably calculated when the user himself is not speaking, since his voice would also be picked up by the microphone positioned close to his ear, and might have a detrimental effect on the performance of the biometric characteristics determination unit. Any suitable technique for determining whether or not the user is speaking can be applied to determine when to switch between the two microphones.

The aural biometric characteristics of a user might be used to identify the user before granting him access to, for example, an online banking service operating from a remote device. To this end, the aural biometric characteristics might be transmitted for verification to a remote device. Since a user's aural biometric characteristics are to be used for identification and/or for granting him access to applications to which only he should have access, the aural biometric characteristics are preferably encrypted before being transmitted. Alternatively, the user's aural biometric characteristics might be stored locally in the audio device. For example, the aural biometric characteristics of the owner of a mobile phone might be stored locally in the phone, authorising only that person to use that phone, and thus offering an effective deterrent against mobile phone theft.

If the audio device is being implemented as, for example, a mobile phone, a derivative of the second microphone signal can also be transmitted to a remote device or "far end". Here, the term "derivative" simply means that a microphone signal is not transmitted directly from the microphone, but after being processed in some way. The derivative of the second microphone signal is obtained by performing AEC on the user speech signal.

The relative positions of the first and second microphones, and the fact that both microphones pick up the same background noise, albeit to different degrees, can be used to good effect in reducing the noise component of the speech signal. Since the second microphone is favourably positioned to pick up the user's speech, the second microphone signal, which is intended for transmission to a far end, will comprise a relatively greater

proportion of speech than of noise. On the other hand, the first microphone is positioned further away from the mouth, so that the first microphone signal will comprise a relatively greater proportion of noise. Therefore, in a further preferred embodiment, these two signals can be processed together in a signal conditioning unit to improve the signal-to-noise ratio, and therefore the quality, of the speech signal prior to transmission.

An appropriate method of identification of a user on the basis of aural biometric characteristics of that user comprises an enrolment step in which a reference set of aural biometric characteristics of a user are determined as described above, and stored in a storage device to enrol a user for access to an application. In a following identification step, an identification set of aural biometric characteristics of a user are determined as described above, and compared to the reference set of aural biometric characteristics. Here, the comparator may make use of known techniques of pattern recognition to find a reference set to match the identification set of aural biometric characteristics. Identification of the user succeeds if a discrepancy between the identification set of aural biometric characteristics and the reference set of aural biometric characteristics falls below a pre-defined confidence measure.

A "confidence measure" is essentially a measure of the accuracy of the system. For some applications, a very high level of accuracy is required, whereas a lower level might suffice for other applications. The accuracy of the system can be expressed in terms of false acceptance ratio (FAR), where a user is erroneously identified or accepted, and false rejection ratio (FRR), in which a user is erroneously rejected. The audio device, comparator and decision making unit might be realised as a single device. However, since the realisation of a comparator and/or decision making unit might require more space than might be available in a user-friendly audio device, these comparator and decision making units might be realised separately, at the same or at different locations.

A corresponding system for biometric user identification comprises an audio device as described above for determining an identification set of aural biometric characteristics of a user, and an interface or access unit for accessing a reference set of aural biometric characteristics of the user, determined and stored, for example, in a previous enrolment step. The system further comprises a comparator to compare or match the feature vector of the reference set to the feature vector of the identification set, and a decision making unit for deciding whether any discrepancy between the identification set of aural biometric characteristics and the reference set of aural biometric characteristics falls below a pre-defined confidence measure.

The storage device used for storage of the identification set of aural biometric characteristics can be, for example, a database on a computer, a SMART card, a SIM card, or any suitable memory device.

For example, the owner of a mobile phone might enrol himself as the authorised user for this mobile phone. To this end, he might initiate an enrolment step by which his aural biometric characteristics are calculated and stored for use in a later verification step, either locally on the SIM card of the mobile phone, or remotely after transmission of the aural biometric characteristics to a remote device or application. The owner of the mobile phone might wish to authorise a further person, or further persons, to use his mobile phone. He might do this by initiating an enrolment step and allowing the person to be authorised to hold the mobile phone to his ear, in order for his aural biometric characteristics to also be stored for use in a future verification step. The mobile phone might include software for checking or verifying whether a person using the phone is also one of the authorised persons. This verification might be carried out before a user of the mobile phone even makes a connection. Therefore, such an application of the invention offers very effective protection of mobile phone theft, since only the authorised persons will be able to use the mobile phone to make calls.

Other objects and features of the present invention will become apparent from the following detailed descriptions considered in conjunction with the accompanying drawings. It is to be understood, however, that the drawings are designed solely for the purposes of illustration and not as a definition of the limits of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will now be described in more detail, by way of example, with reference to the accompanying drawings, wherein:

Fig. 1 is a schematic diagram of an audio device according to a first embodiment of the invention,

Fig. 2 is a more detailed block diagram of the audio device according to the first embodiment of the invention,

Fig. 3 is a block diagram of an audio device according to a second embodiment of the invention,

Fig. 4 is a block diagram of an audio device according to a third embodiment of the invention,

Fig. 5 is a block diagram of system for biometric user identification according to the

invention together with an application, and

Fig. 6 is a schematic diagram of a realisation of an audio device according to an embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

In the drawings, like numbers refer to like objects throughout.

Fig. 1 shows a simplified representation of an audio device 1, which might be a mobile phone, telephone handset, headset, etc. The audio device comprises a loudspeaker 2 and microphone 4 and receives an input signal S_i which is ultimately to be converted to an acoustic signal S_L by the loudspeaker 2. The loudspeaker 2 is positioned close to the ear 3 of a user in such a way that the acoustic signal S_L, originating in the loudspeaker 2, is directed into the ear canal of the ear 3. For the sake of simplicity, the diagram does not show any output signals, which will be considered in more detail below.

The acoustic loudspeaker signal S_L is reflected back out of the ear canal in the form of an aurally reflected signal S_A. Both the loudspeaker signal S_L and aurally reflected signal S_A are picked up, or recorded, by the microphone 4. Furthermore, any other acoustic signals – besides the loudspeaker signal S_L and aurally reflected signal S_A – collectively regarded as a background noise signal S_N originating from a figurative source of noise 17, are also picked up by the microphone 4. A microphone input signal M therefore comprises a loudspeaker component, an aurally reflected component, and a background noise component.

In its simplest form, the audio device 1 comprises two signal processing blocks 5, 6. One of these, an active noise cancellation unit 5, serves to reduce the background noise component of the microphone signal M by effectively subtracting it from the input signal S_i to give a modified loudspeaker output signal, which, when combined with the actual background noise, effectively cancels out the influence of the background noise as perceived by the user.

The other signal processing block, an adaptive filter arrangement 6, processes the loudspeaker and aurally reflected components of the microphone signal M to determine biometric characteristics of the ear canal, and ultimately of the user.

Fig. 2 shows the composition of the active noise cancellation unit 5 and the adaptive filter arrangement 6 in more detail.

The active noise cancellation unit 5 comprises a transfer function block 7, which performs a transfer function H_{ls2mic} on the input signal S_i. The transfer function applied is that of the loudspeaker to microphone, giving effectively the same signal that would be recorded

by the microphone if no other signal components were picked up. The output of the transfer function block is subtracted from the microphone input signal M in a summation unit 15. The output of this summation unit 15, comprising basically everything but the loudspeaker signal S_L – i.e. the background noise S_N – is forwarded to an inverse transfer function block 8 and a subsequent low pass filter 9 to modify the noise component S_N, giving an "anti-noise" signal component which is subtracted from the incoming signal S_i to give a modified "anti-noise" version S_i' of the input signal S_i. This "anti-noise" input signal S_i' is converted by the loudspeaker 2 into an acoustic signal S_L that effectively cancels out the background noise, as might be perceived by a listener.

The adaptive filter arrangement 6 comprises a filter unit 10, an update block 11 and a summation unit 16. The filter unit 10 receives the modified input signal S_i', performs a further modification, and outputs a signal S_X. The summation unit 16 subtracts the signal S_X from the microphone input signal M and passes the result to the update block 11. On the basis of the summation result, the update block 11 continually adjusts the coefficients of the filter unit 10, until they arrive at a good estimate of the loudspeaker to microphone transfer function H_{ls2mic}. This transfer function H_{ls2mic} characterizes the aural biometric of the user, and can therefore be used for reliable user identification. Techniques for deriving such transfer functions are known and described in the literature, for example in H. Moller et al., "Transfer Characteristics of Headphones measured on Human Ears", J. Audio Eng. Soc., Vol. 43, 4, 1995.

Since the filter 10 of the adaptive filter arrangement 6 is adapted continually until it approximates a filter performing the loudspeaker to microphone transfer function H_{ls2mic}, and, on the other hand, a loudspeaker to microphone transfer function H_{ls2mic} is required in the active noise cancellation unit 5, the adaptive filter arrangement 6 can preferably be used to improve the performance of the active noise cancellation unit 5. This is shown in Fig.3, which is basically the same as Fig.2 with an additional control signal 22. This control signal 22 "informs" the transfer function block 7 and the inverse transfer function block 8 of the active noise cancellation unit 5 of the "correct" filter performance. The transfer function block 7 and the inverse transfer function block 8 can then be modified accordingly, thereby improving the performance of the active noise cancellation unit 5. Generally the adaptive filter arrangement 6 is realized using digital components. These digital components can be used to control the ANC circuit 5 which is realized generally using analog components. The technique of using digital controllers to modify the performance of analog cancellation electronics is known, and has also been described in US 5,440,642.

Delays may be incurred by the elements of the active noise cancellation unit 5 and the adaptive filter arrangement 6 during signal processing for the determination of the aural biometric characteristics. To compensate for these delays, appropriate delay elements 12, 13 can be inserted as required into the microphone output signal M before the summation units 15, 16.

The audio device 1 according to the invention can easily be realised in the form of a telephone handset, headset, or mobile phone, as illustrated in Fig. 4. Here, the configuration described in Fig. 2 and Fig. 3 is augmented by a second microphone 14 for picking up a speech signal S_V issuing from the 19 mouth of the user. The adaptive filter arrangement 6 can also be used to reduce unwanted echo effects – perceived as detrimental to the quality of a telephone conversation – by connecting it between the second microphone 14 and the loudspeaker 2. Since aural biometric characteristics are only calculated when the user is not speaking, both first microphone 4 and second microphone 14 can share the single adaptive filter arrangement 6. A switch 34 for switching one of the microphone signals through to the adaptive filter arrangement 6 can be software controlled, so that the first microphone 4 is routed to the adaptive filter arrangement 6 whenever the user is not talking, or when a connection is being set up, or at any other appropriate time. At any other time, the voice input signal M_V picked up by the second microphone 14 can be switched through to the adaptive filter arrangement 6 and modified to give a speech microphone output signal 37, free of any loudspeaker contribution.

The modified speech microphone output signal 37 is routed to a signal conditioning unit 30 before transmission as an output signal 38 to a far end, where it is used appropriately. For example, the output signal 38 might be transmitted to a remote audio device such as a telephone, and there converted into audible sound by the telephone loudspeaker. A data signal 36, comprising the aural biometric characteristics of the ear 3 of the user, might be sent to an appropriate analysis unit for performing the necessary steps of user enrolment and/or user verification. Such an analysis unit, not shown in the diagram, might be realised in the audio device, or externally, depending on the intended application and on device realisation constraints.

Fig. 5 illustrates a system 40 for biometric user identification, where an audio device 1, comprising a loudspeaker 2 and microphone 4, is used to determine aural biometric characteristics for a user in the manner described above, on the basis of a loudspeaker signal S_L and the resulting aurally reflected signal S_A reflected from the ear canal of the user's ear 3.

In a first enrolment step, a reference set A_{ref} of aural biometric characteristics is

determined and transmitted to a remote device 41, which might be in the vicinity of the audio device 1 or at a remote location. Transmission might be effected via a cable link, or in a wireless manner. The reference set A_{ref} of aural biometric characteristics is forwarded by an interface 42 to a suitable storage device 43 for storage.

At a later point in time, in an identification step, an identification set A_{id} of aural biometric characteristics is generated in the audio device 1, and is also transmitted to the remote device 41, where it is forwarded to a comparator 44. The interface 42, operating in this step as an accessing unit 42, retrieves the reference set A_{ref} of aural biometric characteristics from the storage device 43 and forwards these to the comparator 44. The comparator 44 compares both sets A_{id}, A_{ref} of aural biometric characteristics using appropriate techniques of pattern matching or feature vector analysis. The results of the comparison are passed to a decision making unit 45, which determines whether the match is sufficiently close to warrant a positive identification of the user. The decision making unit 45 reports its decision – "accept" or "reject" – in the form of an appropriate identification signal 46 to the application to which the user is attempting to gain access.

The application can use the identification signal 46 to determine whether it should continue or abort. For example, a positively identified user, attempting to access a telephone brokerage service for which he has previously enrolled by allowing his reference set A_{ref} of aural biometric characteristics to be stored, is indicated by a corresponding "accept" identification signal 46. The telephone brokerage service can assume that the user is who he claims to be, and responds to the user's wishes, transmitted perhaps as spoken commands over the telephone connection. In the case of a negative decision of the decision making unit and a subsequent "reject" identification signal 46, the telephone brokerage service can report the failure to the user attempting to gain access, via the telephone connection, and terminate the connection.

Fig. 6 shows a possible embodiment of the audio device 1. Here, the audio device 1 comprises a typical mobile phone. The loudspeaker 2 is positioned so that it can be held close to the ear whilst maintaining a suitable distance between the speaker microphone 14 and the mouth of the user. A microphone 4 for use in active noise cancellation and biometric characteristic determination is positioned close to the loudspeaker 2. This illustration demonstrates that the functionality of the methods according to the invention can be realised in an audio device of the type commonly available.

Although the present invention has been disclosed in the form of preferred embodiments and variations thereon, it will be understood that numerous additional

modifications and variations could be made thereto without departing from the scope of the invention. For example, the filters used in the adaptive filter arrangement, acoustic echo cancellation circuit and active noise cancellation circuit can be analog filters, digital filters, or a combination of both. A reference set of aural biometric characteristics for use in user identification can be determined in a different manner to that described above.

For the sake of clarity, it is also to be understood that the use of "a" or "an" throughout this application does not exclude a plurality, and "comprising" does not exclude other steps or elements. A "unit" may comprise a number of blocks or devices, unless explicitly described as a single entity.

CLAIMS

1. An audio device (1) comprising:

- a loudspeaker (2) for outputting a loudspeaker output signal (S_L) close to the ear (3) of a user of the audio device (1);
- a first microphone (4) arranged in a manner suitable for recording a first microphone input signal (M) including an aurally reflected signal component (S_A);
- an active noise cancellation circuit (5) for actively reducing any background noise component (S_N) of the microphone input signal (M);
- a biometric characteristics determination unit (6) for determining the aural biometric characteristics of the user on the basis of the noise-reduced microphone input signal (M).

2. An audio device (1) as claimed in claim 1, wherein the biometric characteristics determination unit (6) comprises an adaptive filter arrangement (10) capable of performing the functions of acoustic echo cancellation.

3. An audio device (1) as claimed in claim 1, comprising a control signal connection between the biometric characteristics determination unit (6) and the active noise cancellation circuit (5) such that a control signal (22) generated by the biometric characteristics determination unit (6) controls the behaviour of the active noise cancellation circuit (5).

4. An audio device (1) as claimed in claim 1, comprising a second microphone (14) arranged in a manner suitable for recording a second microphone input signal (M_V) including a user speech signal component (S_V) for transmission.

5. An audio device (1) as claimed in claim 1, comprising an acoustic echo cancellation arrangement (16) for reducing any loudspeaker output signal component (S_L) of the second microphone input signal (M_V).

6. An audio device (1) as claimed in claim 1, comprising a switch (34) for switching an adaptive filter arrangement (6) between the first microphone input signal (M) and the second microphone input signal (M_V).

7. An audio device (1) as claimed in claim 1, comprising a signal conditioning unit (30) for improving, according to the first microphone signal (M), the quality of a derivative (37) of the second microphone input signal (M_V) prior to transmission.

8. A system (40) for biometric user identification, comprising

- an audio device (1) as claimed in claim 1, for determining an identification set (A_{id}) of aural biometric characteristics of a user;
- an access unit (42) for accessing a reference set (A_{ref}) of aural biometric characteristics of the user;
- a comparator (44) for comparing the identification set (A_{id}) of aural biometric characteristics with the reference set (A_{ref}) of aural biometric characteristics;
- a decision making unit (45) for deciding whether any discrepancy between the identification set (A_{id}) of aural biometric characteristics and the reference set (A_{ref}) of aural biometric characteristics falls below a pre-defined confidence measure.

9. A method of determining biometric characteristics of a user, wherein:

- a loudspeaker output signal (S_L) is emitted from a loudspeaker (2) close to the ear (3) of a user;
- a microphone input signal (M), including an aurally reflected signal component (S_A), is recorded by a first microphone (4);
- a background noise signal component (S_N) of the microphone input signal (M) is reduced by an active noise cancellation circuit (5) to give a noise-reduced microphone input signal (M); and
- aural biometric characteristics of the user are determined using the noise-reduced microphone input signal (M).

10. A method of identification of a user on the basis of aural biometric characteristics of that user, wherein:

- in an enrolment step, a reference set (A_{ref}) of aural biometric characteristics of a user are determined, which aural biometric characteristics are comparable to aural biometric characteristics determined according to the method as claimed in claim 9, and the reference set (A_{ref}) of aural biometric characteristics of the user are stored in a storage device (43);
- in an identification step, an identification set (A_{id}) of aural biometric characteristics of a user are determined according to the method as claimed in claim 9, which identification set

(A_{id}) of aural biometric characteristics is compared to the reference (A_{ref}) set of aural biometric characteristics, and identification of the user succeeds if a discrepancy between the identification set (A_{id}) of aural biometric characteristics and the reference set (A_{ref}) of aural biometric characteristics falls below a pre-defined confidence measure.

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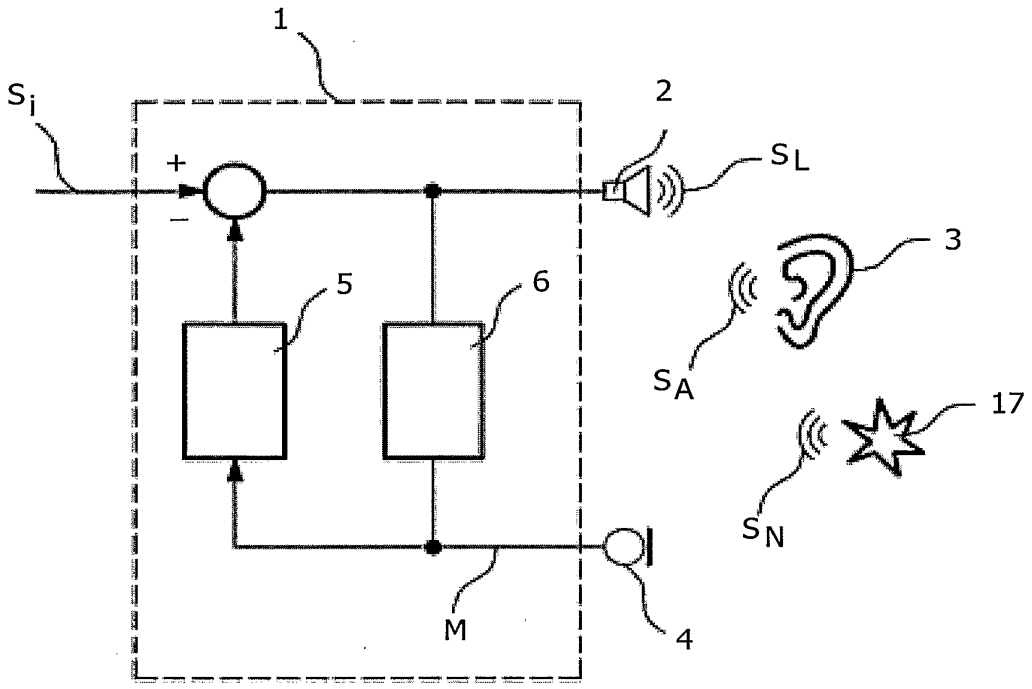


FIG. 1

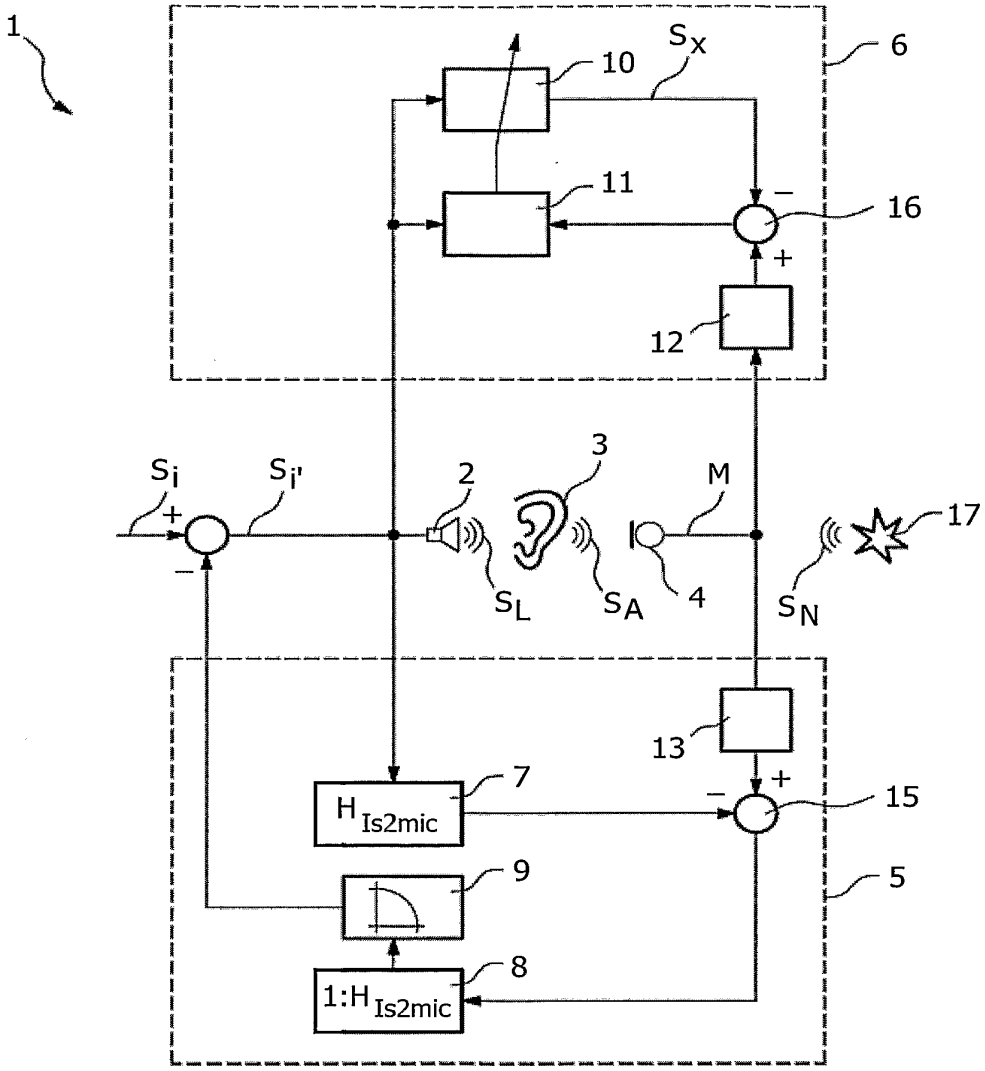

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FIG. 2

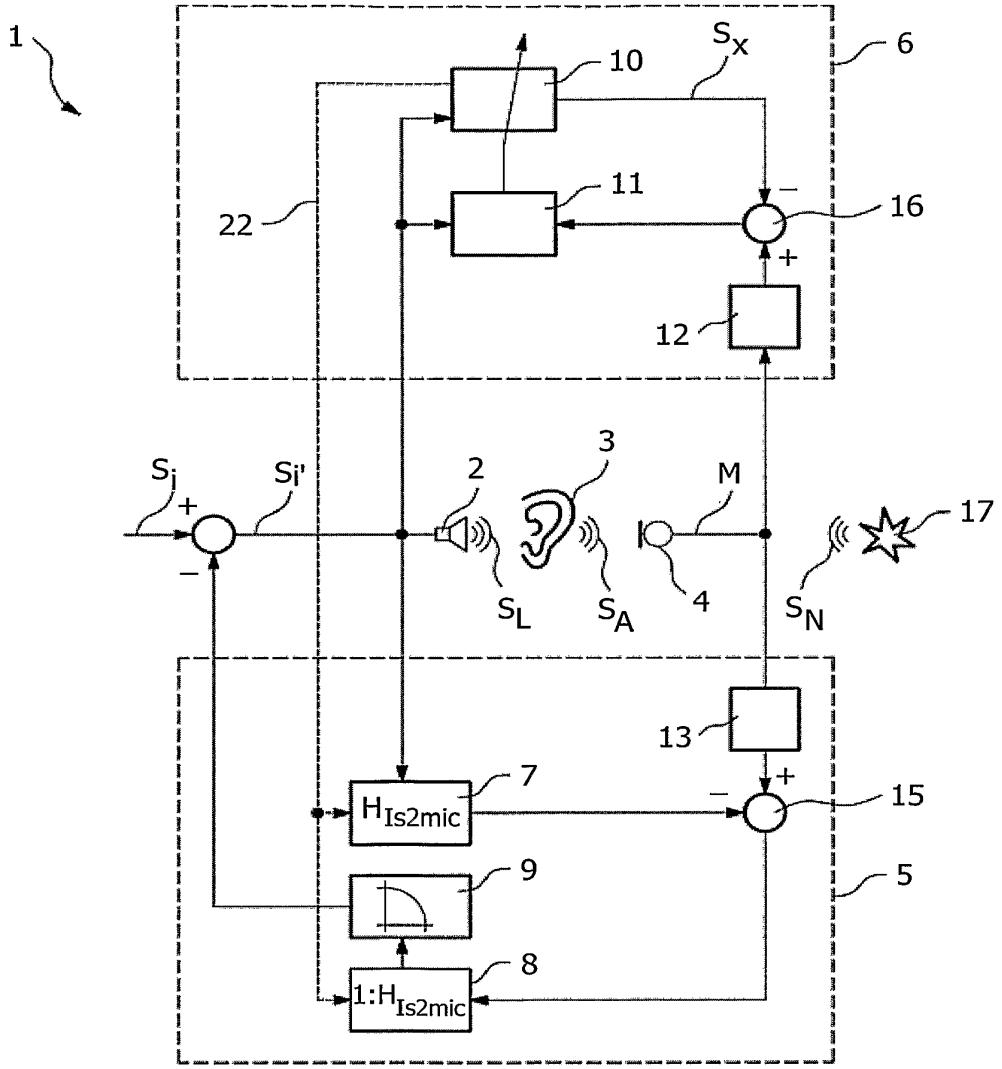

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FIG. 3

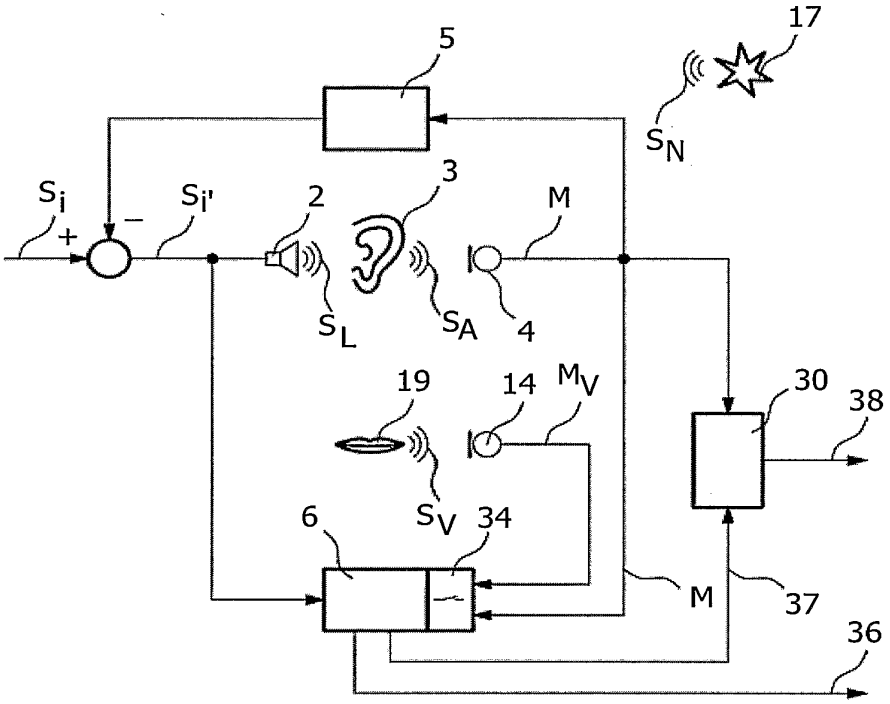


FIG. 4

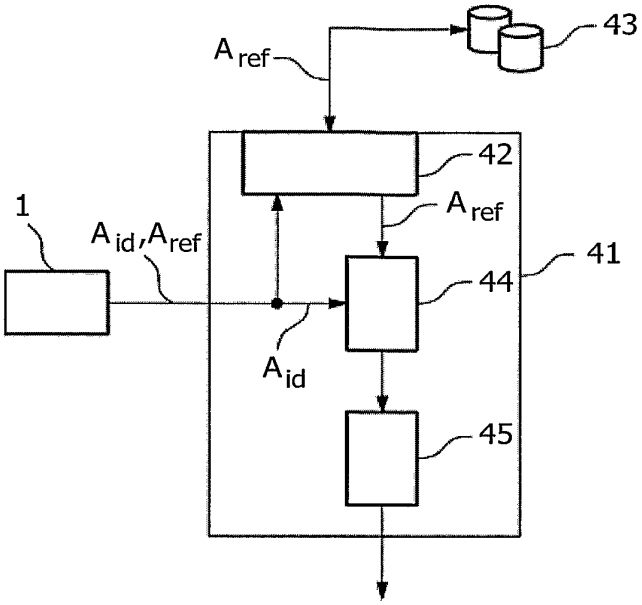


FIG. 5

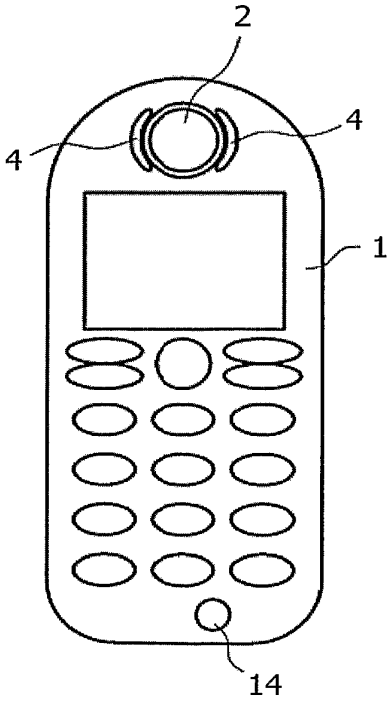


FIG. 6

INTERNATIONAL SEARCH REPORT

International Application No
PCT/IB2005/053686

A. CLASSIFICATION OF SUBJECT MATTER
G06K9/00 A61B5/117

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)
G06K A61B G07C

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

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 02/091310 A1 (ORANGE PERSONAL COMMUNICATIONS SERVICES LTD) 28 October 2004 (2004-10-28) abstract page 18, line 18 - page 19, line 3	1-10

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

☒ Patent family members are listed in annex.

° Special categories of cited documents :

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

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

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

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

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

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

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Date of the actual completion of the international search	Date of mailing of the international search report
20 January 2006	30/01/2006

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Sonius, M

Form PCT/ISA/210 (second sheet) (January 2004)

INTERNATIONAL SEARCH REPORT

nformation on patent family members

International Application No
PCT/IB2005/053686

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 02091310 A1	14-11-2002	EP 1388127 A1	11-02-2004
		GB 2375205 A	06-11-2002
		JP 2004532584 T	21-10-2004
		US 2004215968 A1	28-10-2004

Form PCT/ISA/210 (patent family annex) (January 2004)

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01.07.2005 US 173355

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(54) Method and system for active noise cancellation

(57) A method and system for active noise cancellation is provided. The system employs subband processing, and preferably implements over-sampled filterbank. The system is applicable to adaptive noise cancellation, adaptive echo cancellation for portable listening devices, such as headsets and other similar listening devices.

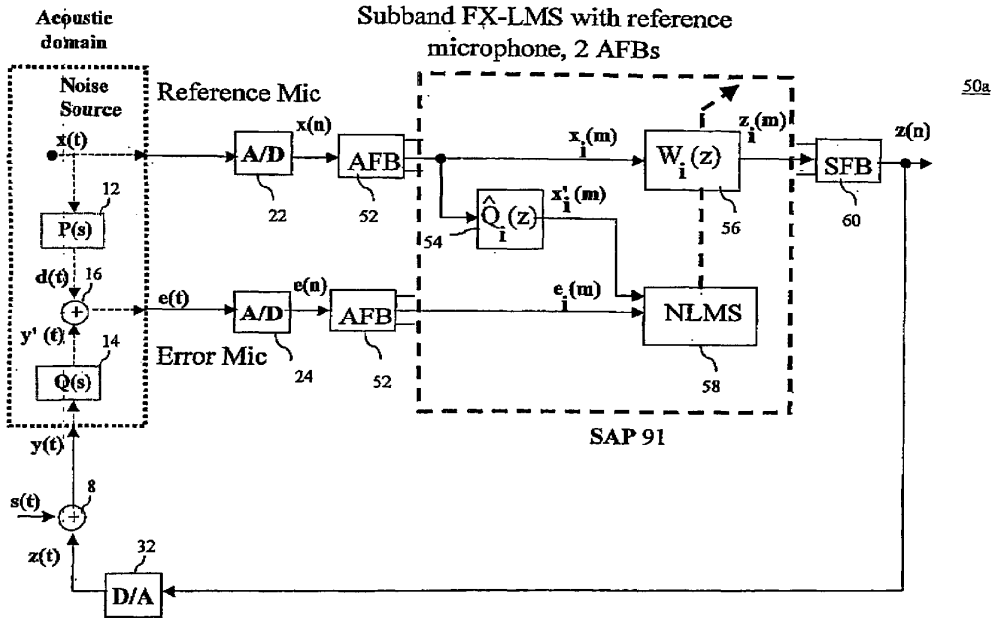


Fig. 4

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EP 1 638 079 A2

Description

FIELD OF INVENTION

[0001] The present invention relates to signal processing and modeling technique, and more specifically to signal processing and modeling technique for noise cancellation.

BACKGROUND OF THE INVENTION

[0002] Analog active noise cancellation (ANC) systems suffer from a number of problems. Specifically, they are prone to acoustic feedback, and they do not provide as high a degree of cancellation for periodic or other quasi-stationary signals as can be realized with a digital signal processing (DSP) enhanced analog ANC system.

[0003] Analog ANC systems are also difficult to adjust (or "tune") for different headset designs and also in a production environment where normal production variations in transducers and listening device assembly increase the likelihood of acoustic feedback.

[0004] Further, current analog ANC techniques address only part of the noise cancellation that is needed by users in high noise environments. Specifically, analog ANC provides noise cancellation at predominantly low frequencies (below 1500 Hz to 2000 Hz),

[0005] Fully digital ANC systems are possible. However, group delay or latency is induced by analog to digital (A/D) conversion, digital to analog conversion and digital processing associated with DSP systems. Further, due to power consumption, they are not practical in many portable applications.

[0006] U.S. Patent Nos. 5,475,761, 5,699,436, and 5,815,582 by Noise Cancellation Technologies (NCT) disclose methods of digital ANC using a combination of both feedback and feed-forward methods. The methods employ DSP to perform ANC, However, due to the inherent delay in the DSP, they are not practical for most applications when low-power, low-cost, and small-size constraints are applied. There are many other similar DSP-based systems that suffer from the same delay problem, for example, the systems disclosed in U.S. Patent Nos. 6,418,227 B1, 5,991,418 and 5,940,519 by Kuo et al. from Texas Instruments Inc.

[0007] U.S. Patent Nos. 6,069,959 and 6,118,878 byNCT disclose fully analog solutions to the ANC problems. Specifically, as U.S. Patent No. 6,118,878 explains, significant tuning and adaptation of the system parameters are necessary to avoid instability and artifacts. However, the patent suggests that the tuning can be implemented using analog components and methods.

[0008] DSP-controlled ANC systems have tried to address the difficult problem of tuning of the analog ANC systems through the use of CPUs and signal processing methods. For example, U.S. Patent 5,440,642 by Denenberg et al. discloses DSP techniques that can control ANC system parameters, such as loop gain and loop filter frequency response. U.S. Patent Application Publication No. 20040037430 A1 uses DSP techniques (LMS adaptation) to control the secondary path typically used in the filtered-X LMS algorithm. U.S. Patent No. 4,965,832 uses DSP control of a feed-forward ANC system to control the loop-gain and the loop-filter bandwidth. U.S. Patent No. 6,278,786 B1 uses DSP to not only control the loop-gain but also to provide an acoustic signal (to be added to the analog ANC anti-noise) that will cope with tonal noises more effectively.

[0009] Subband adaptive filters (SAFs) become an interesting and viable option for many adaptive systems. The SAF approach uses a filterbank to split the fullband signal input into a number of frequency bands, each serving as input to an adaptive filter. This subband decomposition greatly reduces the update rate and the length of the adaptive filters resulting in much lower computational complexity.

[0010] To be able to employ powerful SAF method for ANC, one has to tackle a processing delay issue.

[0011] To reduce the processing delay in the SAFs, U.S. Patent No. 5,329,587 by Morgan et al, has introduced a method of reconstructing the subband filter back into time-domain. Starting with adapted subband filters, they first transform the SAFs into the frequency-domain (using an FFT), appropriately stack the results, and inverse transform them back into time-domain to obtain a time-domain adaptive filter. The time-domain filter is then used to implement time-domain adaptive filtering. The details of their technique are also reported in a research paper Morgan et. al. ("A delayless subband adaptive filter structure", IEEE Trans. on Signal Proc., Vol. 43, pp. 1819-1830, Aug. 1995.) that offers a good survey of previous efforts on low-delay adaptive systems. Let us call this method DFT-1 Stacking as disclosed in J. Huo et al. ("New weight transform schemes for delayless subband adaptive filtering", in Proc. of IEEE Global Telecom. Conf., pp. 197-201,2001). After analyzing Morgan's method in by J. Huo et al., they offer two variations to the method (known as "DFT-2 Stacking" and "DFT-FIR Stacking") to improve the performance. These methods are all based on DFT, proper stacking, and inverse DFT. In DFT-FIR, a convolution with a synthesis filter after DFT is also added. Moreover in L. Larson et al, ("A new subband weight transform for delayless subband adaptive filtering structures", in Proc. of IEEE DSP workshop, pp. 201-206), a Linear Weight Transform method is introduced. The method employs a linear matrix transformation of the subband filters using both analysis and synthesis filters to recover the time-domain adaptive filter. In yet another set of works following Morgan's method, a different method is proposed that employs the Hadamard transform to reconstruct the time-domain filter (N. Hirayama and H. Sakai, "Analysis of a delayless subband adaptive filter", in Proc, of ICASSP, pp. 2329-2332,1997; and N, Hirayama et al., "Delayless subband adaptive filtering using the hadamard transform", IEEE Trans. on Signal Proc., Vol. 47, No, 6, pp.

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I731-1734, Jun. 1999).

[0012] In a series of research paper presented from 1997 to 1999, Merched et al. present methods of transferring the SAFs to time-domain (R. Merched et al. "A Delayless alias-free subband adaptive filter structure", in Proc. of IEEE Int. Symp. On Circuits and Systems, pp. 2329-2332, Jun. 9-12,1997; P. S. R. Diniz et al. "Analysis of a delayless subband adaptive filter structure", in Proc. of ICASSP, pp. 1661-1664, 1998; R. Merched et al. "A new delayless subband adaptive filter structure", IEEE Trans. on Signal Proc., Vol. 47, No. 6, pp. 1580-1591, Jun. 1999). Their methods are designed only for maximally decimated (QMF) PR filter banks, and constraints the filterbank prototype filter to be a Nyquist(K) filter (where K represents number of subbands). As a result, the SAFs become simple fractional delay filters. They also use a polyphase fiterbank to reconstruct the time-domain adaptive filter.

[0013] U.S, Patent No. 6,661,895 B1 by Jin et al. discloses a zero-delay SAF system. They discard the initial segment of each SAF to obtain a "forward filter". The estimated (time-domain) echo signal is generated by filtering the reference signal through the subband forward filters and then applying subband reconstruction. The time-domain echo cancelled signal goes through another separate time-domain LMS filter to compensate for the discarded initial segments of subband adaptive filters. The method however has a fundamental problem: the time-domain LMS filter has to model a potentially non-causal filter. This is not practically possible.

[0014] Over-sampled subband adaptive filters (OS-SAF) offer many advantages over time-domain adaptive algorithms. However, OS-SAF systems may introduce a group delay or latency that is too high for some applications. The conversion from analog to digital and back again, use of anti-aliasing and anti-imaging filters, as well as the digital processing introduces this delay, Reducing the delay of OS-SAF systems by increasing the sampling rate is not practical in many applications because of power consumed and specialized hardware required.

[0015] Also, in the conventional OS-SAF systems, the primary input signal goes through analysis and synthesis stages of the over-sampled filterbank. Often, perfect reconstruction (PR) is not practical and only a near PR performance is achieved. As a result, the primary signal may be distorted. To minimize distortions, longer analysis windows have to be employed which further increases the delay and add extra computation cost.

[0016] Further, in the conventional OS-SAF systems, the effect of analysis filter band edges and under-modeling errors limit the system performance.

SUMMARY OF THE INVENTION

[0017] It is an object of the invention to provide a novel method and system that obviates or mitigates at least one of the disadvantages of existing systems.

[0018] It is an object of the invention to provide an improved method and system for active noise cancellation.

[0019] The invention relates to the improvements that can be made to well-known analog ANC techniques as well as over-sampled subband adaptive filtering using specialized DSP methods and apparatuses.

[0020] According to an aspect of the present invention, there is provided a system for active noise cancellation, includes: a first over-sampled analysis filterbank for transferring a reference signal in a time-domain, which is associated with noise, into a plurality of subband reference signals in a frequency-domain; a second over-sampled analysis filterbank for transferring a primary signal in the time-domain, which is associated with an acoustic signal and may be contaminated by the noise, into a plurality of subband primary signals in the frequency-domain; a subband processing module for processing the subband reference signals, the subband primary signals or a combination thereof, and implementing one or more than one subband adaptive algorithm in the frequency-domain; and an over-sampled synthesis filterbank for transferring the outputs of the subband processing module into a time-domain output signal.

[0021] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes: an over-sampled analysis filterbank for transferring a primary signal in a time-domain, which is associated with an acoustic signal and may be contaminated by noise, into a plurality of subband primary signals in a frequency-domain; a subband processing module for the subband primary signals and implementing one or more than one subband adaptive algorithm in frequency-domain; and an over-sampled synthesis filterbank for transferring the outputs of the subband processing module into a time-domain output signal.

[0022] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes; a first analysis filter bank for transferring a reference signal in a time-domain into a plurality of subband reference signals in a frequency-domain; a second analysis filter bank for transferring a primary signal in the time-domain, which is associated with an acoustic signal and may be contaminated by noise, into a plurality of subband primary signals in the frequency-domain; a subband estimator for modeling subband acoustic transfer function for the subband reference signals; a subband adaptive filter for providing a plurality of subband output signals in response to the subband reference signals; an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the subband reference signals; and a synthesis filter bank for transferring the subband output signals to a time-domain output signal.

[0023] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes: an analysis filter bank for transferring a primary signal in a time-domain into a plurality of subband primary signals in a frequency-domain; a subband

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filter bank for providing a plurality of subband output signals in response to a plurality of subband reference signals; a synthesis filter bank for transferring the subband output signals into a time-domain output signal; and a feed-back loop for generating the subband reference signals, including: a first subband estimator for modeling subband acoustic transfer function for the subband output signals; a signal path for providing the subband reference signals in response to the subband primary signals and the modeling for the subband output signals; a second subband estimator for modeling subband acoustic transfer function for the subband reference signals; and an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the subband reference signals.

[0024] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes: a first analysis filter bank for transferring a primary signal in a time-domain into a plurality of subband primary signals in a frequency-domain; a time-domain filter bank for providing a time-domain output signal in response to a reference signal in the time-domain; a feed-back loop for generating the reference signal, including: a first subband estimator for modeling subband acoustic transfer function for the time-domain output signal, a signal path for providing the reference signal in the time-domain in response to a primary signal in the time-domain and the modeling for the time-domain output signal, a second subband estimator for modeling subband acoustic transfer function for the subband reference signal, a second analysis filter bank for transferring the primary signal into a plurality of subband primary signals in the frequency-domain, an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the reference signal, and a synthesis filter bank for converting the subband adaptive filter to the time-domain filter bank for filtering the reference signal.

[0025] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes: an analog active noise cancellation (ANC) system for performing an active noise cancellation to a primary signal in a time-domain, which is associated with an acoustic signal and may be contaminated by noise; a first over-sampled analysis filterbank for transferring a reference signal in the time-domain into a plurality of subband reference signals in a frequency-domain, the reference signal in the time-domain being associated with the noise; a second over-sampled analysis filterbank for transferring the primary signal in the time-domain into a plurality of subband primary signals in the frequency-domain; a subband processing module for processing the subband reference signals, the subband primary signals or a combination thereof, and for adjusting one or more parameters of the analog ANC system; an over-sampled synthesis filterbank for performing conversion on the outputs of the subband processing module from the frequency-domain to the time-domain.

[0026] According to a further aspect of the present invention, there is provided a system for active noise cancellation, includes: an analog active noise cancellation (ANC) system for performing an active noise cancellation to a primary signal in a time-domain, which is associated with an acoustic signal and may be contaminated by noise; an over-sampled analysis filterbank for transferring the primary signal in the time-domain into a plurality of subband primary signals in a frequency-domain; a subband processing module for processing the subband primary signals and for adjusting one or more than one parameter of the analog ANC system; an over-sampled synthesis filterbank for transferring the outputs of the subband processing module into an output signal in the time-domain.

[0027] According to a further aspect of the present invention, there is provided a system for active noise cancellation, comprising: a first WOLA analysis filterbank for transferring a reference signal in a time-domain, which is associated with noise, into a plurality of subband reference signals in a frequency-domain; a second WOLA analysis filterbank for transferring a primary signal in the time-domain, which is associated with an acoustic signal and may be contaminated by the noise, into a plurality of subband primary signals; a subband adaptive processing module for processing the output of the first WOLA analysis filterbank, the output of the second WOLA analysis filterbank or a combination thereof, and providing a plurality of subband adaptive filters; and a WOLA synthesis filterbank for synthesizing the subband adaptive filters to provide a time-domain filter for filtering the reference signal in the time-domain.

[0028] According to a further aspect of the present invention, there is provided a method for active noise cancellation implemented by the systems described above.

[0029] This summary of the invention does not necessarily describe all features of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

[0030] These and other features of the invention will become more apparent from the following description in which reference is made to the appended drawings wherein:

[0031] FIGURE 1(a) is a diagram showing a conventional analog ANC system;

[0032] FIGURE 1(b) is a diagram showing a detailed block diagram of Figure 1(a), depicting a transfer function and an acoustic transfer function;

[0033] FIGURE 2 is a diagram showing a conventional bSP-based ANC system;

[0034] FIGURE 3 is a diagram showing a subband ANC system using a subband FX-LMS;

[0035] FIGURE 4 is a diagram showing a subband ANC system using a subband FX-LMS in accordance with an embodiment of the present invention;

[0036] FIGURE 5 is a diagram showing a conventional feedback ANC system using a subband FX-LMS;

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[0037] FIGURE 6 is a diagram showing a subband feedback ANC system using a subband FX-LMS in accordance with a further embodiment of the present invention;

[0038] FIGURE 7 is a diagram showing a delayless subband ANC system using a FX-LMS in accordance with a further embodiment of the present invention;

[0039] FIGURE 8 is a diagram showing a delayless subband feedback ANC system using a FX-LMS in accordance with a further embodiment of the present invention;

[0040] FIGURE 9 is a diagram showing an ANC system using over-sampled subband processing in accordance with a further embodiment of the present invention;

[0041] FIGURE 10 is a diagram showing a feedback ANC system using over-sampled subband processing in accordance with a further embodiment of the present invention;

[0042] FIGURE 11 is a diagram showing an ANC system using Weighted Overlap-Add (WOLA) in accordance with a further embodiment of the present invention;

[0043] FIGURE 12 is a diagram showing a feedback ANC system using WOLA in accordance with a further embodiment of the present invention;

[0044] FIGURE 13 is a diagram showing an ANC system using WOLA in accordance with a further embodiment of the present invention;

[0045] FIGURE 14 is a diagram showing a conventional OS-SAF system;

[0046] FIGURES 15(a)-(b) are diagrams showing examples of an adaptive processing block (APB);

[0047] FIGURE 16 is a diagram showing an OS-SAF having WOLA;

[0048] FIGURE 17 is a diagram showing an OS-SAF system using over-sampled subband processing in accordance with a further embodiment of the present invention;

[0049] FIGURE 18 is a diagram showing an OS-SAF system using over-sampled subband processing in accordance with a further embodiment of the present invention;

[0050] FIGURE 19 is a diagram showing an OS-SAF system using WOLA in accordance with a further embodiment of the present invention;

[0051] FIGURE 20 is a diagram showing a closed-loop OS-SAF system using over-sampled subband processing in accordance with a further embodiment of the present invention;

[0052] FIGURE 21 is a diagram showing a closed-loop OS-SAF system using WOLA in accordance with a further embodiment of the present invention;

[0053] FIGURE 22 is a diagram showing a delayless subband feedback ANC system using WOLA in accordance with a further embodiment of the present invention;

[0054] FIGURE 23 is a diagram showing a delayless SAF system using oversampled filterbanks, employing weight transform for time-filter reconstruction;

[0055] FIGURE 24 is a diagram showing a Reconstruction of TAF through WOLA synthesis of the SAFs in accordance with a further embodiment of the present invention;

[0056] FIGURE 25 is a diagram showing details of time-filter reconstruction using the WOLA process;

[0057] FIGURE 26 is a diagram showing an oversampled SAF system applied to echo cancellation;

[0058] FIGURE 27 is a graph for showing the simulation results of Figures 24-26;

[0059] FIGURE 28 is a graph showing the ERLE results;

[0060] FIGURE 29 is a graph showing an example of real and imaginary parts of SAFs;

[0061] FIGURE 30 is a graph showing an example of periodic extension of a WOLA synthesis;

[0062] FIGURE 31 is a graph showing an example of a synthesis window;

[0063] FIGURE 32 is a graph showing the result of widow application of Figure 31;

[0064] FIGURE 33 is a graph showing an example of time sample for synthesizing a time-domain filter;

[0065] FIGURE 34(a) is a graph showing an example of time sample for a synthesized time-filter super-imposed on the ITUT plant;

[0066] FIGURE 34(b) is a graph showing an example of time-domain difference between the synthesized time-filter and the ITUT plant; and

[0067] FIGURE 35 is a graph showing echo attenuation of a time-domain filter and a conventional SAF.

DETAILED DESCRIPTION

[0068] The embodiments of the present invention are described for a headset. However, the embodiments of the present invention are applicable to any other listening devices, such as portable listening devices. The embodiments of the present invention are described mostly for active noise cancellation and echo cancellation. However, the embodiments of the present invention can be employed for other applications, including adaptive noise cancellation.

[0069] The embodiments of the present invention relate to over-sampled subband adaptive filtering using specialized DSP techniques and analog ANC techniques. Thus, DSPs are relevant to the technology disclosed below. Because the applications of these techniques are in listening devices and the cancellation relies on acoustic summation, the embodiments of the present invention relates to acoustics.

[0070] Figure 1(a) illustrates a conventional analog ANC system 2 for a headset, As shown in Figure 1(a), a primary noise signal x(t) is sensed by a microphone 6. The microphone 6 is usually located within the earcup of the headset. For example, the primary noise signal x(t) is a signal outside the earcup of the headset. An analog ANC circuitry 4 receives the microphone signal e(t), and generates an electric signal z(t). The electric signal z(t) is added at 8 with a local audio signal s(t) (possibly

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speech) to generate an electric speaker signal y(t). The electric speaker signal y(t) is played through a loudspeaker 10 for the listener. The loudspeaker 10 is located within the earcup of the headset, The ANC system 2 tries to cancel the effect of a noise signal for the listener through estimating, generating the signal z(t) to be played through the loudspeaker 10 together with the local audio signal s(t).

[0071] Figure 1(b) illustrates modeling of a transfer function and an acoustic transfer function of Figure 1(a). P(s) 12 models the transfer function for the acoustic noise signal x(t) to be converted to an electric signal d(t). Q(s) 14 models the acoustic transfer function for the loudspeaker signal y(t) to reach the microphone (6). Usually Q(s) 14 is assumed to be more known than P(s) 12 since the locations of the loudspeaker (10) and the microphone (6) are fixed and known. A through review of the conventional analog ANC system 2 is provided in Kuo-Morgan99 (Sen M. Kuo and Dennis R. Morgan, "Active Noise Control: A Tutorial Review", Proceedings of the IEEE, VoL 87, June 1999, pp. 943-973).

[0072] ANC systems may also provide one or more microphones to measure the ambient noise (e.g. signal x(t) outside of the earcup), however, single microphone systems are generally preferred as discussed later.

[0073] Figure 2 illustrates a conventional DSP-based ANC system 20. The system 20 of Figure 2 is a feed-forward ANC system, and employs the FX LMS algorithm for active noise cancellation with two microphones. The two microphone signals x(t), e(t) are converted to digital signals x(n), e(n) by analog/digital (A/D) converters 22 and 24, and processed in discrete-time by an algorithm to generate an anti-noise signal z(n) . The anti-noise signal z(n) is converted back to an analog signal z(t) by a digital/analog (DIA) converter 32, and played through the loudspeaker together with the signal s(t). The method might employ adaptive methods, such as the Normalized Least Mean Square (1VLMS) 30 or similar techniques to adapt an adaptive filter W(z) 28. A rough estimate of the loudspeaker to the error microphone transfer function Q (s) is also required. In Figure 2, this is depicted by the discrete-time estimated transfer function Q(z) 26. Various methods for off-line on-line estimation of Q(s) have been proposed in the prior art and reviewed in Kuo-Morgan99.

[0074] Subband ANC methods have been presented in Kuo-Morgan99 to achieve lower computation cost and faster convergence. Figure 3 illustrates a conventional subband ANC system for two microphones, employing a subband FX-LMS,

[0075] The system 40 of Figure 3 includes three Analysis Filter Bank (AFB) components 42. The AFB components 42 decompose the time-domain signals e(n), x(n), x'(n) into K (possibly complex) subband signals $e_i(m)$ $x_i(m)$, $x_i'(m)$, i=0,1,...,K-1 that might be also decimated in time. There exist K (possibly complex) subband adaptive filets (SAFs) Wi(z) 44 ($W_i(z)$, i=0,1,..., K-1) which generate subband output signals $z_i(m)$, i=0,1,...,K-1. All of the adaptive processing is done in a subband adaptive processing (SAP) block 90 in Figure 3. A Synthesis Filter Bank (SFB) 46 then combines the subband output signals to obtain the time domain signal z(n). The D/A converter 32 converts the time domain, digital signal z(n) into a time domain, analog signal z(t).

[0076] Figure 4 illustrates a subband ANC system 50a for two microphones in accordance with an embodiment of the present invention. In Figure 4, adaptive processing is implemented in a SAP block 91. The system 50a of Figure 4 employs suband FX-LMS, and includes a block 54 that includes a subband estimate of Q(s) depicted as $\hat{Q}_i(z)$,i=0,1,..., K-1. Subband estimation and implementation of Q(s) allows for faster computation due to parallel subband processing by filters $\hat{Q}_i(z)$. This allows the system 50a to include only two AFBs 52 for two microphones. One AFB is provided to x(n), while the other is provided to e(n). K (possibly complex) subband adaptive filers (SAFs) $W_i(z)$ 56 ($W_i(z)$, i=0,1,..., K-1) generate subband output signals $z_i(m)$, i=0,1,...,K-1, based on $x_i(m)$, i=0,1,..., K-1. A block (such as NLMS) 58 is provided to adapt the subband adaptive filters Wi(z), A SFB 60 combines the subband output signals $z_i(m)$, i=0,1,...,K-1, to obtain the time domain signal z(n), The D/A converter 32 converts the time domain, digital signal z(n) into a time domain, analog signal z(t).

[0077] It is possible to implement FX-LMS with only one microphone as illustrated in Figure 5. Figure 5 shows a conventional feedback ANC system 70. The system 70 is disclosed in Kuo-Morgan99. In Figure 5, the reference signal is reconstructed in the system (signal r(n)) via a discrete-time estimated transfer function $\hat{Q}(z)$ 72 and a summation node 74,

[0078] The system 50a of Figure 4 may be implemented using oversampled filterbank as shown in Figure 9, AFBs 52 of Figure 4 may be implemented by over-sampled analysis filterbanks 112,114 of Figure 9, WOLA implementation offers a low-delay, flexible, and efficient implementation of the over-sampled filterbanks as described in U.S. Patent Nos. 6,236,731, 6,240,192 and 6,115,478, which are incorporated herein by reference. The system 50a of Figure 4 may be implemented using WOLA filterbank as shown in Figures 11 and 16. AFBs 52 of Figure 4 may be implemented by WOLA analysis filterbanks 132, 134 of Figures 11 and 16, and SFB 60 of Figure 4 may be implemented by a WOLA synthesis filterbank 13 8 of Figures 11 and 16.

[0079] Figure 6 illustrates a subband feedback ANC system 50b in accordance with a further embodiment of the present invention. A subband implementation of feedback FX-LMS system is shown in Figure 6. In Figure 6, adaptive processing is implemented in a SAP block 92. The reference signal is reconstructed in the system 50b via $\hat{Q}_i(z)$, i=0,1,..., K-1 (referenced by 80) and a summation node 82. The block 80 includes a subband estimate of Q(s) depicted as $\hat{Q}_i(z)$, i=0,1,..., K-1.

[0080] As discussed in the prior art (Kuo-Morgan99), the use of AFBs in subband implementations may impose

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delays on the signal that are often prohibitively large for the operation of the system.

[0081] The system 50b of Figure 6 may be implemented using oversampled filterbank as shown in Figure 10. AFB 52 of Figure 6 may be implemented by an over-sampled analysis filterbank 114 of Figure 10. The system 50b of Figure 6 may be implemented using WOLA filterbank as shown in Figure 12, AFB 52 of Figure 6 may be implemented by a WOLA analysis filterbank 134 of Figure 12, and SFB 60 of Figure 6 may be implemented by a WOLA synthesis filterbank 138 of Figure 12,

[0082] Delayless subband ANC systems associated with the systems of Figures 4 and 6 are shown in Figures 7 and 8. Figure 7 illustrates a delayless feed-forward subband ANC system 50c in accordance with a further embodiment of the present invention. Figure 8 illustrates a delayless subband feedback ANC system 50d in accordance with a further embodiment of the present invention. In Figure 7, adaptive processing is implemented in a SAP block 93, In Figure 8, adaptive processing is implemented in a SAP block 94. Each of the systems 50c and 50d includes the components of the system 50a of Figure 4, and further includes a single time-domain filter W(z) 84. The time-domain filter W(z) 84 is an FIR adaptive filter synthesized from subband adaptive filters 56 by SFB 60 and applied for adaptive filtering in time-domain.

[0083] In each of Figures 7 and 8, SFB 60 is provided to convert the SAFs $W_i(z)$, i=0,1,..., K-1 into the single time-domain filter W(z)84. Thus, delays due to AFBs are not seen in the signal path. The method to obtain the time-filter from the SAFs is disclosed below in a further embodiment(s) associated with delayless SAF. Using this method, the processing delay of the filterbank is eliminated from the adaptive processing. As a result, more non-stationary components of the noise can also be cancelled through the digital ANC part of the system.

[0084] The system 50c of Figure 7 may be implemented using oversampled filterbank as shown in Figure 17. AFBs 52 of Figure 7 may be implemented by over-sampled analysis filterbanks 112, 114 of Figure 17. The system 50c of Figure 7 may be implemented using WOLA filteibank as shown in Figures 13 and 19. AFB 52 of Figure 7 maybe implemented by WOLA analysis filterbanks 132,134 of Figures 13 and 19, and SFB 60 of Figure 7 may be implemented by a WOLA synthesis filterbank 138 of Figures 13 and 19,

[0085] The system 50c of Figure 8 may be implemented using oversampled filterbank. The system 50c of Figure 8 may be implemented using WOLA filterbank as shown in Figure 22. AFBs 52 of Figure 8 may be implemented by WOLA analysis filterbanks 132,134 of Figure 22, and SFB 60 of Figure 8 may be implemented by a WOLA synthesis filterbank 138 of Figure 22.

[0086] U.S. Patent Application Publication No. 20030198357 (Serial No. 10/214,056), entitled "Sound Intelligibility Enhancement Using a Psychoacoustics Model and an Over-sampled Filterbank", which is incorporated herein by reference, discloses the use of ANC in combination with other techniques to improve the intelligibility of audio signals. The sound intelligibility enhancement disclosed in this U.S. application is applicable to the ANC systems of Figures 3, 4 and 6-8,

[0087] Convergence improvement techniques such as whitening by decimation (WBD), whitening by spectral emphasis (WBS), and whitening by decimation and spectral emphasis (WBDS), disclosed in U.S. Patent Application Publication Nos. 20030108214 and 20040071284 (Serial Nos. 10/214,057 and 10/642,847), which are incorporated herein by reference, can be employed in combination with all methods and systems described in Figures 3, 4, and 6-8.

[0088] A combination of an analog ANC and subband processing is now described in detail. In Figures 9-13, the analog ANC and the subband processing are combined to achieve a higher performance as described below.

[0089] Figure 9 illustrates an ANC system 100a in accordance with a further embodiment of the present invention. The system 100a of Figure 9 includes an analog ANC 105 and subband processing. The analog ANC 105 may be the analog ANC 2 of Figure 1.

[0090] Comparing Figure 9 with Figure 1 (b), it can be seen that the analog ANC system is entirely embedded in the system 100a. The system 100a further includes a second (Reference) microphone that is possibly located outside of the headset earcup to sample the noise. The two microphone signals (x(t), e(t)) are converted to digital, discrete-time signals by the A/D converters 22 and 24 to obtain the signals x(n) and y(n) . The signals are next processed by two (identical) over-sampled analysis filterbanks 112 and 114. The subband processing block 116 processes the over-sampled subband signals ($x_i(m)$ and $y_i(m)$, i=0,1,...,K-1) output from the over-sampled analysis filterbanks 112 and 114, and detects various system and signal conditions including the brink of instability of the analog ANC 105. Accordingly, it can tune and adjust parameters of the analog ANC system 105 (such as loop-gain, and loop-filter bandwidth) and/or turn on or off certain features (such as feedback loop) of the system 100a. The outputs of the subband processing block 116 are synthesized by an over-sampled synthesis filterbank 118, which produces the time domain, digital signal z(n).

[0091] The embodiments of the present invention may also employ over-sampled subband processing to provide improved cancellation of periodic or other quasi-stationary signals. Here, the ambient noise (measured by a reference where a microphone is possibly outside the earcup for a feedforward system) is analyzed using subband techniques to determine if there are any stationary (ideally periodic) or quasi-stationary components in the ambient noise. If these components are detected, the DSP will generate a delayed version of this stationary or quasi-stationary signal (shown as z(n) in digital form and z(t) in analog) in Figure 9 and supply it to the analog ANC subsystem (105) for subtraction.

[0092] Adaptive techniques, such as subband adap-

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tive filters on over-sampled filterbanks (OS-SAFs) similar to those disclosed in U.S, Patent Application Publication Nos. 20040071284 and 20030108214 can be employed. The FX-LMS algorithm may be employed in subband methods described in Figures 3, 4, and 6-8, where OS-SAFs and SAPs are employed. This method of combining the analog and digital ANC provides improved noise cancellation compared to a system that does not employ this technique.

[0093] It is noted that in Figure 9, the reference microphone could have dual usages: 1) to provide information to the subband processing block about the ambient noise in order to control the parameters of the analog ANC system, 2) to provide a reference signal for digital ANC part of the system (employing algorithms such as the FX-LMS). In the system 100a of Figure 9, various versions of the subband FX-LMS, such as feed-forward, feedback, and a combination of the two, may be used for the digital ANC part of the system.

[0094] In some systems, only one microphone is available (such as feedback ANC). The system 100a of Figure 9 can be modified to obtain another embodiment disclosed in Figure 10. Figure 10 illustrates a feedback ANC system 100b in accordance with a further embodiment of the present invention. In the system 100b, the subband signals for controlling the analog ANC system 105 are provided only by the error microphone. The SAP block 92 of Figure 6 can be employed as a part of the subband processing block 116 of Figure 10 to do adaptive processing. Similarly, the SAP block 91 of Figure 4 can be employed in the subband processing block 116 of Figure 9,

[0095] The over-sampled filterbanks may be efficiently implemented using WOLA analysis and synthesis. Figure 11 illustrates an ANC system 100c in accordance with a further embodiment of the present invention. The system 100c of Figure 11 includes WOLA analysis filterbanks 132 and 134 and a WOLA synthesis filterbank 138. Efficient hardware realizations of the WOLA have been disclosed in U.S. Patent Nos. 6,236,731 and 6,240,192.

[0096] The embodiments of Figures 7 and 10 may be efficiently implemented using the WOLA filterbanks as shown in Figures 12 and 13, respectively. Figure 12 illustrates a feedback ANC system 100d in accordance with a further embodiment of the present invention. Figure 13 illustrates an ANC system 100e in accordance with a further embodiment of the present invention. The subband processing block 116 of Figure 12 may employ feed-forward FX-LMS strategies, such as the SAP block 94 of Figure 8. The subband processing block 116 of Figure 13 may employ feed-forward FX-LMS strategies such the SAP block 93 of Figure 7.

[0097] The subband processing block 116 of Figures 9-13 may model one or more than one acoustic transfer function, a transfer function for a microphone, a transfer function for a loudspeaker, or combinations thereof in accordance with an application,

[0098] Monitoring the amplitude to the noise to be cancelled may save the battery life. The over-sampled filterbank can also perform this monitoring more accurately using subband processing, since more accurate decision making is possible by monitoring energies of the signals in various subbands. For example, different energy thresholds may be employed for different subbands according to the effectiveness of ANC in various frequency bands, '

[0099] Delayless SAP through over-sampled synthesis/WOLA synthesis is now described in detail.

[0100] Figure 14 illustrates a conventional OS-SAF system 140a. The OS-SAF system 140a includes the over-sampled analysis filterbanks 112 and 114, subband adaptive processing blocks (APBs) 142, and the over-sampled synthesis filterbank 118, The OS-SAF system 140a has two inputs, i.e., a primary input e(n) (e,g., the error microphone signal in the ANC systems of Figures 2-5), and a reference input x(n) (e.g., the reference signal in the ANC systems of Figures 2-5). The reference inputs leaks into the primary input for example by going through an acoustic plant P(s) in Figures 2-5, As a result, the primary and reference inputs become correlated. The OS-SAF system 140a tries to eliminate the portion of the primary input that is correlated with the reference input through adaptive filtering.

[0101] Figures 15(a)-15(b) illustrate examples of adaptive processing. The APBs of Figures 15(a)-15(b) are applicable to APB 142 of Figure 14. The APB of Figure 15(b) contains a summation node 148, while the APB of Figure 15(a) lacks the summation node 148.

[0102] The APB of Figure 15(a) is applicable to the ANC applications, such as the FX-LMS method of Figures 2-6. The summation node 146 is rather transferred to the acoustic domain as shown in the ANC systems of Figures 2-6. The APB of Figure 15(b) is applicable to applications, such adaptive interference (echo or noise) cancellation. Here the interference signal is cancelled in the digital domain through the adaptive algorithms 144, Both of the two APBs in Figures 15(a)-(b) could be equally used in the embodiment of the present invention disclosed here. For brevity, it is assumed that in the description below, the APB 142 has the form of Figure 15(b), unless otherwise stated.

[0103] An example of the disclosed time-filter reconstruction through WOLA synthesis is described for an echo cancellation application as follows.

[0104] As disclosed in U.S. Patent Application Publication Nos. 20030108214 and No. 20040071284 (Serial Nos. 10/214,057 and 10/642,847), which are incorporated herein by reference, the over-sampled analysis and synthesis filterbank operations can be efficiently implemented using the WOLA analysis and synthesis, respectively, WOLA analysis/synthesis for oversampled filterbank analysis/synthesis is disclosed in U.S. Patent Nos. 6,236,731 and 6,240,192, which are incorporated herein by reference. Figure 16 illustrates an OS-SAF system 140b having WOLA analysis filterbanks 132 and 134 and a WOLA synthesis filterbank 138.

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[0105] Figure 17 illustrates an OS-SAF system 150a in accordance with a further embodiment of the present invention. In the system 150a, the subband adaptive filters $W_k(n)$ are combined together through the filterbank synthesis process to obtain a time-domain adaptive filter W(z) 154 of W(n). The adaptive filter W(z) 154 receives the reference input x(n). The output of the adaptive filter W(z) 154 and the reference input e(n) are summed at 156.

[0106] Figure 18 illustrates an OS-SAF system 150b in accordance with a further embodiment of the present invention. In the AFBs 142 of the system 150b, the AFB of Figure 15(b) is employed.

[0107] Figure 19 illustrates an OS-SAF system 150c in accordance with a further embodiment of the present invention. The system 150c corresponds to the system 150a of Figure 17, and utilizes the WOLA filterbank. As illustrated in Figure 19, the WOLA synthesis could be used to efficiently synthesize the time-domain adaptive filter. The WOLA synthesis process includes steps disclosed in U.S. Patent Nos. 6,236,731, 6,240,192 and 6,115,478.

[0108] Closed-loop versions af the OS-SAF systems disclosed above are also possible. Figure 20 illustrates a closed-loop version of the system 150a in Figure 17, and Figure 21 discloses a closed-loop version of the system 150c in Figure 19. The feedback systems 150d and 150e of Figures 20-21 offer steady-state performance, since the final error signal is sensed back by the system and optimally eliminated.

[0109] Figure 22 illustrates a delayless WOLA-based subband feedback ANC system using a FX-LMS in accordance with a further embodiment of the present invention. The system 150h of Figure 22 corresponds to the system 50d of Figure 8, and utilizes the WOLA filterbank.

[0110] A. Filter reconstruction for Low-resource delayless subband adaptive filter using WOLA

[0111] Method and system for reconstruction of a time-domain adaptive filter (TAF) by using WOLA synthesis method (implemented with an oversampled filterbank) is described in detail. An inverse fast Fourier transform (IFFT) of length K is employed in the method. Due to the nature of the WOLA synthesis, the reconstruction process is distributed in time rendering it suitable for real-time implementation. The method is arranged such that segments of the TAF may be used as they become available in time. This makes the method a perfect match for sequential partial update algorithms (described later) that are often integral parts of low-resource implementations. The WOLA synthesis of the subband filters described below is efficiently implemented on an oversampled filterbank that also benefits from the WOLA implementation for its analysis stage. The system is designed and described for an echo cancellation set up thongh it could be used for other adaptive applications such active noise cancellation or adaptive feedback cancellation,

[0112] A-1. WOLA Synthesis for Filter Reconstruction

[0113] Figure 23 illustrates a delayless SAF structure with echo plant. At the output of APBs of Figure 23, the sub-band adaptive filters $P_k(z)$, $k=0,1,\cdots, K-1$ are obtained. Rather than reconstructing the output signal as in typical SAF systems, the adaptive filters are passed to a weight transformation stage to obtain the time-domain adaptive filter $P(z)$ to be used to filter the reference signal $x(n)$ in the time-domain. Assuming a synthesis filter set, in the DFT-FIR approach the TAF is obtained by passing the SAFs through a synthesis filterbank as described by the J. Huo reference and the L. Larson reference:

[0114]

$$P(z) = \sum_{k=0}^{K-1} F_k(z) P_k(z^R)\, z^{-Ls/2}$$

[0115] where R is the filterbank decimation factor, and $F_0(z)$ is the prototype filter of the filterbank, bandlimited to π/R. Synthesis filters are obtained through discrete Fourier transform (DFT) modulation of the prototype filter as $F_k(z)=F_0(zW^k)$ where $W=e^{-j2\pi/K}$. The term $z^{-Ls/2}$ is added to compensate for delay of the synthesis filter of length Ls.

[0116] The DFT-FIR uses a polyphase-FFT structure to reconstruct the TAF through a weight transform of the SAP set. By contrast in this section, a different approach is described whereby the TAF is reconstructed through a WOLA synthesis of the SAFs. This method is more amenable to a block processing approach and more compatible to hardware implementation. Basically, this method treats the SAFs $p_k(m)$, $k=0,1,\cdots, K-1$, $m=0,1,\cdots, M-1$ as a set of K subband signals, and passes them through an oversampled filterbank synthesis stage as shown in Figure 24. To efficiently implement the oversampled synthesis stage, we use the WOLA implementation as depicted in Figure 25. For further explanation, consider the SAFs all included in an SAF matrix P, with elements defined as $P(k,m)= P_k(m)$, $m=0,1,\cdots,M-1$, $k=0,1,\cdots,K-1$. The matrix is set as input to the synthesis stage, one column at every subband time-tick. We call this method of TAF reconstruction "sequential synthesis". As depicted in Figure 25, the WOLA synthesis starts with taking an IFFT of each column (of length K) of the SAF matrix. After the IFFT 166, and proper circular shifting 168, the vectors of length K are periodically extended to obtain a vector as long as the synthesis window. Next this result is multiplied by the synthesis window followed by an overlap-add operation. Both evenly-stacked FFT and oddly-stacked FFTs may be used. Odd stacking requires an extra sign sequencer to be employed at the final stage. The WOLA synthesis is well described in U.S. Patent Nos. 6,236,731, 6,240,192 and 6,115,478. Assuming aliasing is low in the analysis stage, the SAFs can be shown to converge to the Wiener solution. As a result,

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the solution will be almost independent of the analysis filter design. Thus the synthesis filter set $F_k(z)$, = *0,1,···, K-1* is designed independently of the analysis filter set to constitute a near perfect-reconstruction set. To obtain the TAF, the output buffer 168 in Figure 25 is first zeroed out. After reading in the input SAF matrix (one column per subband clock tick), the first *Ls/2* samples of the output are discarded. The next *L* output samples produced a block at a time (*R* time-samples) constitute the TAF. Thus it takes *L/R* input (subband) clock ticks to obtain the TAF. Through optimuzed implementation it became possible to avoid the initial *Lsl2* sample delays between consecutive filter reconstructions. The total input-output delay for the TAF filter reconstruction is thus *(La + Ls)/2* samples where La denotes the analysis window length. This delay is not seen in the signal path; rather the optimal filter for the reference and primary signals is delayed by this amount. When the echo plant varies slowly (relative to this reconstruction delay), this delay does not degrade the system performance. It is possible to minimize the delay by choosing shorter analysis and synthesis windows as long as distortions in the time-filter due to the reconstruction process are kept within a tolerable range.

[0117] In the conventional SAF systems shorter analysis/synthesis windows will lead to increased output signal degradation since all signals pass through the complete filterbank as described in H. Sheikhzadeh et al. ("Performance Limitations of a New Subband Adaptive System for Noise and Echo Reduction", *Proc. Of IEEE ICECS,* 2003). By contrast, in the delayless SAF system of Figure 24, signals are passed through the analysis filterbank only to obtain the adaptive filter. As a result, the adverse effects of shorter analysis/synthesis windows on output signal quality is much less pronounced. It is also possible to further reduce the TAF reconstruction delay to only *La/2* samples if one is ready to perform filterbank reconstruction of the whole SAF matrix P for every output block. We call this method of TAF reconstruction "batch synthesis" as opposed to the "sequential synthesis" described in this section. Batch synthesis will increase the computation cost from a single WOLA synthesis (per output block) in the sequential synthesis to *M*WOLA synthesis operations. It is also possible to sequentially reconstruct more than one (but less than *M*) columns of matrix P at every subband time-tick if the computational resources permit.

[0118] A-2 System Evaluation

[0119] We evaluate the WOLA filter reconstruction process for a filterbank set up of: analysis and synthesis window lengths of *La = 64* , and *Ls =128* samples with *K = 16* subbands and decimation rate of *R = 4*. The analysis filter is shorter in length (and wider in frequency domain) compared to the synthesis filter. This was chosen to provide better excitation in the analysis filter transition region leading to better convergence behavior as reported by P. L. De Leon II et al ("Experimental Results with Increased Bandwidth Analysis Filters in Oversampled, Subband Acoustic Echo Cancellers," *IEEE Sig. Proc. Letters,* Jan , 1995, vol. 2, pp 1-3). Each SAF p_k (*m*) is of length *M=32* resulting in a *16* × *32* SAF matrix P . For comparison, the same filterbank setup was also employed for the oversampled SAF system with the WOLA implementation depicted in Figure 26. The echo plant was the eighth plant of the ITUT G. 168 standard (Recommendation ITU-T G.168, Digital Network Echo Cancellers, Int'1 Telecommunication Union, 2000), the Echo Return Loss (ERL) was *10* dB, and random white noise was used at the reference input without any near-end disturbance.

[0120] For subband adaptation, the Gauss-Seidel Pseudo-Affine Projection (GS-PAP) (Albu et al., "The Gauss-Seidel Pseudo Affine Projection Algorithm and its Application for Echo Cancellation," *37th Asilomar Conf. Sig. Sys. &Comp.,* Pacific Grove, Calif., Nov. 2003) with an affine order of two was employed. The method provides fast convergence and is simple enough to be targeted for a low-resource real-time implementation. GS-PAP may be superior to the Fast Affine Projection Algorithm (FAPA) since unlike the FAPA that operates on a transformed adaptive filter, the GS-PAP directly provides the adaptive filter itself (the Albu et al. reference). To demonstrate the capability of the WGLA filter reconstruction algorithm in matching sequential update algorithms, sequential update GS-PAP (SGS-PAP) described in H. Sheikhzadeh et al. ("Partial update subband implementation of complex pseudo-affine projection algorithm on oversampled filterbanks", ICASSP, 2005 IEEE International Conference on Acoustics, Speech, and Signal Processing, March 18-23, 2005, US) was also used for subband adaptation.

[0121] SGS-PAP employs sequential update of the adaptive filter in the framework of GS-PAF to reduce computation cost. The sequential decimation factor of the SGS-PAP was chosen to be eight (*D=8*). This means only one polyphase component (of length *32/8=4* taps) out of a total of *8* components of each SAF is adapted at each subband clock tick. For *D=1*, the S GS-PAP is obviously the same as GS-PAP. In the delayless algorithm, a block of *R* new samples of the TAF is available every subband tick. This new block is used to update the TAF as soon as it becomes available. This way a smooth and continuous filter reconstruction is achieved.

[0122] Figure 27 shows (A) Time-domain echo plant, and the reconstructed plant with SGS-PAP adaptation, for *D=1,8,* and (B) error in reconstructed TAFs for *D=1* and (C) error for *D=8*

[0123] Figure 27 illustrates the ERLE results for the SAF system of Figure 26 as well as the delayless WOLA filter synthesis algorithm (Figures 24 and 25), employing SGS-PAP with *D=1,8*. As expected, the delayless method achieves a greater ERLE compared to the SAF system, partially due to the fact that the input signals are not passed through the WOLA analysis/synthesis stages. This is consistent with the result and analysis presented in the J, Huo reference and the L. Larson reference. The SGS-PAP method shows a slight performance degrada-

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tion for *D=8* (for both the SAF and the delayless methods) since the subband adaptive filters are updated at a much lower rate. Adaptation cost is, however, reduced by a factor of *D=8* .

[0124] Figure 28 illustrates the ERLE results for SGS-PAP adaptation with *D=1,8* , for the SAF system of Figure 26 and the delayless SAF algorithms of Figure 24.

[0125] Using the WOLA synthesis, the TAF was synthesized. Figure 27-A shows the synthesized time-filter for SGS-PAP adaptation with *D=1,8*, super-imposed on the ITUT plant. As shown the three impulse responses are almost identical. To observe the differences, Figures 27-B and 27-C depict time-domain differences between each of the synthesized time-filters and the ITUT plant for *D=1,8*. As shown, the differences are negligible in both cases, and higher for *D=8* as predicted by the ERLE results.

[0126] The WOLA structure is more amenable with a block processing system (R.E. Crochiere et al. "Multirate digital signal processing" , Prentice-Hall, NJ, 1983). As illustrated in Figure25, for every input vector, all the processes (after the IFFT) occur sequentially to generate *R* samples of output. All various blocks of Figure 25 can thus operate synchronously with a single subband clock, and there is no need to extra buffering. Since the WOLA filterbank is a block processing system, every slice generated from the IFFT block that must be overlap-added to the previous results has a common exponent. This simplifies the architecture, reducing power and enhancing throughput as described in U.S. Patent Nos. 6,236,731, 6,240,192 and 6,115,478. The same does not hold for straightforward polyphase implementation of the J. Huo reference since it is a stream processing method. In the polyphase filterbank synthesis of the J. Huo reference, a separate convolution of one of polyphase synthesis filters (with one of the K subband signals) has to occur for each sample of output. To do this, *K* different data buffers have to be updated. To summarize, the WOLA synthesis of Figure 25 provides a simpler, more modular structure for real-time hardware implementation as described in U.S. PatentNos. 6,236,73I, 6,240,192 and 6,115,478.

[0127] The method, based on WOLA synthesis of the SAFs, is efficient and is well mapped to a low-resource hardware implementation. Also the WOLA adaptive filter reconstruction may easily be spread out in time simplifying the necessary hardware. This time-spreading may be easily combined with partial update adaptive algorithms to reduce the computation cost for low-resource real-time platforms.

[0128] A-3 Filter Reconstruction and Evaluation

[0129] An example for reconstruction by WOLA is described, using WOLA setup of analysis window length of L = 128, with K = 32 subband, and decimation rate of R=8. Assume (without loss of generality) that APB's are of the form illustrated in Figure 15(b) employed for an echo cancellation application. The collection of K SAFs for subbands k = 0,1,..., K-1 is shown in Figure 25, Each SAF W_k (m) is of length M,n=0,1,...,M-1 resulting in a Kby M SAF matrix. The system 150c of Figure 19 for echo cancellation is employed using the eighth plant of the ITUT G. 168 standard for echo generation. The reconstruction process of Figure 25 was applied to this simulation.

[0130] Figure 29 shows an example of real and imaginary parts of SAFs for subbands k = 0,1,2,3 for the subband adaptive system after convergence. Treating each SAF as a subband signal, the process of WOLA synthesis starts with taking the IFFT of each column (of length K) of the SAF matrix. There are M columns to be processed. After IFFT, and proper circular shifting, the vectors of length K are periodically extended to obtain vectors as long as the synthesis window. The result of periodic extension is shown in Figure 30 for the SAFs shown in Figure 29, where a total length of 256 samples (8 periodic extensions) is used. Then a synthesis window (of length Ls = 256 samples) is applied to the periodically extended signals. A typical window is shown in Figure 31 and the result of window application is shown in Figure 32. The last step is the overlap-add of the vectors (a total of M vectors involved). Using a properly designed synthesis window, the time-domain filter is synthesized in time as shown in Figure 33. Figure 34(a) shows the synthesized time-filter super-imposed on the ITUT plant As shown, the synthesized time-filter and the ITUT plant are almost identical. Figure 34(b) illustrates the time-domain difference between the synthesized time-filter and the TTUT plant. As shown, the difference is negligible.

[0131] The time-domain filter can be used for adaptive filtering. As illustrated in Figure 35, for random noise input and the same ITUT plant, the echo attenuation of the time-domain filter is superior to the SAF filter using the NLMS algorithm. This is partially due to the fact that the input signal to the time-domain filter is not passed through the WOLA analysis stage (hence not experiencing distortions), as shown in Figure 4.

[0132] It is noted that as for the method of adaptation of SAFs (e.g,, NLMS 30, 58), the Normalized Least Mean Square (NLMS), the Affine Projection Algorithm (APA) and its variants such as Fast APA (FAPA), or the Recursive Least Squares (RLS) may also be used.

[0133] The corresponding Canadian Patent Application No. 2,481,629 also describes reconstruction of TAF by WOLA, which is incorporated herein by reference,

[0134] The embodiments of the present invention reduces acoustic feedback and provide as high a degree of cancellation for periodic or other quasi-stationary signals.

[0135] The embodiments of the present invention automatically adapts to the different acoustic situations presented by different headsets and to the normal variations encountered in production parts.

[0136] As described above, over-sampled filterbank processing and system architecture disclosed in U.S. Patent Nos. 6,240,192 B1, 6,236,731 B1, and 6,115,478 provide low group delay, low power and small size. The

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embodiments of the present invention can be efficiently implemented on the system architecture disclosed in U.S. Patent Nos. 6,240,192 B1, 6,236,731 B1, and 6,115,478.

[0137] As described above, an analog ANC provides noise cancellation at predominantly low frequencies (below 1500 Hz to 2000 Hz), The embodiments of the present invention can extend this frequency range. The introduction of a DSP processor also permits additional processing to be incorporated, including techniques that can improve speech intelligibility at frequencies where even DSP enhanced analog ANC ceases to provide benefit, as described in U.S. Patent Application Publication No. 20030198357 (Serial No. 10/214,056). U.S. Patent Application Publication No. 20030198357 discloses a sound intelligibility enhancement (SIE) using a psychoacoustic model and preferably an oversampled filterbank. When combined with ANC, the SIE performs better since ANC provides more benefit at low frequencies while SIE provides more benefit at high frequencies. Using the ANC , undesired dynamic range reduction is avoidable due to low-frequency noise.

[0138] The embodiments of the present invention disclosed above prevents acoustic feedback in a manner that retains high fidelity and good performance in the presence of typical disturbances. Subband acoustic feedback reduction is employed to extend the operating frequency range by permitting more loop gain to be introduced before the onset of acoustic feedback. This type of feedback cancellation introduces fewer artifacts than full band approaches for feedback cancellation. It also provides better performance in the presence of coloured noise (disturbances). Finally the reduced group delay (compared to a full band system with similar feedback cancellation performance) provides a faster response time for feedback cancellation,

[0139] As detailed in the U.S. Patent No. 6,118,878, the brink of instability can be detected by finding the ratio of external to internal noise at high frequencies. Alternatively, the residual signal can be monitored at various frequencies to detect and prevent an impending instability. Employing an over-sampled filterbank provides more accurate and reliable prediction of impending instabilities. Generally, the overall control of the ANC system employing an over-sampled subband approach is more efficient, and accurate.

[0140] The delayless OS-SAF in accordance with the embodiments of the present invention considerably reduces the delay introduced in the primary signal path. Also, since the actual echo cancellation occurs in time-domain, inevitable distortions due to over-sampled filterbank analysis/synthesis is avoided. As the results demonstrate, the delayless SAF method proposed outperforms the traditional OS-SAFs in terms of adaptive noise and echo cancellation performance. This is possible due to avoiding errors due to band edges in the OS-SAF system.

[0141] The embodiments of the present invention provide a combination of OS-SAF and time-domain filtering. Comparing to a full band system with similar adaptive processing, this combination provides a faster convergence and response time of the adaptive system.

[0142] The embodiments of the present invention may be implemented by hardware, software or a combination of hardware and software having the above described functions. The software code, either in its entirety or a part thereof, may be stored in a computer readable medium. Further, a computer data signal representing the software code which may be embedded in a carrier wave may be transmitted via a communication networlc. Such a computer readable medium and, a computer data signal and carrier wave are also within the scope of the present invention, as well as the hardware, software and the combination thereof.

[0143] All citations are hereby incorporated by reference.

[0144] The present invention has been described with regard to one or more embodiments. However, it will be apparent to persons skilled in the art that a number of variations and modifications can be made without departing from the scope of the invention as defined in the claims.

Claims

1. A system for active noise cancellation, comprising:

   a first over-sampled analysis filterbank for transferring a primary signal in a time-domain, which is associated with an acoustic signal and may be contaminated by noise, into a plurality of subband primary signals in a frequency-domain;
   a subband processing module for the subband primary signals and implementing one or more than one subband adaptive algorithm in the frequency-domain; and
   an over-sampled synthesis filterbank for transferring the outputs of the subband processing module into a time-domain output signal.

2. A system according to claim 1, further comprising:

   a second over-sampled analysis filterbank for transferring a reference signal in the time-domain, which is associated with the noise, into a plurality of subband reference signals in the frequency-domain;
   the subband processing module processing the subband primary signals, the subband reference signals or combinations thereof, and implementing the one or more than one subband adaptive algorithm in the frequency-domain.

3. A system according to claim 1, further comprising:

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an analog active noise cancellation (ANC) system for performing an active noise cancellation to the primary signal in the time-domain,
the subband processing module including means for implementing the adaptive algorithm for adjusting one or more than one parameter of the analog ANC system.

4. A system as claimed in claim 3, comprising:

a second over-sampled analysis filterbank for transferring a reference signal in the time-domain into a plurality of subband reference signals in the frequency-domain, the reference signal in the time-domain being associated with the noise;
the subband processing module processing the subband primary signals, the subband reference signals or combinations thereof, and adjusting the one or more than one parameter of the analog ANC system.

5. A system according to claim 1, wherein

the first over-sampled analysis filterbank includes a first WOLA analysis filterbank for transferring the primary signal in the time-domain, into the plurality of subband primary signals;
the subband processing module includes means for implementing the adaptive algorithm for providing a plurality of subband adaptive filters; and
the over-sampled synthesis filterbank includes a WOLA synthesis filterbank for synthesizing the subband adaptive filters to provide a time-domain filter for filtering a reference signal in the time-domain associated with the noise.

6. A system as claimed in claim 5, comprising:

a second WOLA analysis filterbank for transferring the reference signal in the time-domain, which is associated with the noise, into a plurality of subband reference signals in the frequency-domain;
the subband processing module processing the output of the first WOLA analysis filterbank, the output of the second WOLA analysis filterbank or a combination thereof, and providing the plurality of subband adaptive filters.

7. A system for active noise cancellation, comprising:

a first analysis filter bank for transferring a reference signal in a time-domain into a plurality of subband reference signals in a frequency-domain;
a second analysis filter bank for transferring a primary signal in the time-domain, which is associated with an acoustic signal and may be contaminated by noise, into a plurality of subband primary signals in the frequency-domain;
a subband estimator for modeling subband acoustic transfer function for the subband reference signals;
a subband adaptive filter for providing a plurality of subband output signals in response to the subband reference signals;
an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the subband reference signals; and
a synthesis filter bank for transferring the subband output signals to a time-domain output signal.

8. A system for active noise cancellation, comprising:

an analysis filter bank for transferring a primary signal in a time-domain into a plurality of subband primary signals in a frequency-domain;
a subband filter bank for providing a plurality of subband output signals in response to a plurality of subband reference signals;
a synthesis filter bank for transferring the subband output signals into a time-domain output signal; and
a feed-back loop for generating the subband reference signals, including:

a first subband estimator for modeling subband acoustic transfer function for the subband output signals;
a signal path for providing the subband reference signals in response to the subband primary signals and the modeling for the subband output signals;
a second subband estimator for modeling subband acoustic transfer function for the subband reference signals; and

an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the subband reference signals.

9. A system for active noise cancellation, comprising:

a first analysis filter bank for transferring a primary signal in a time-domain into a plurality of subband primary signals in a frequency-domain;
a time-domain filter bank for providing a time-domain output signal in response to a reference signal in the time-domain;
a feed-back loop for generating the reference signal, including:

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a first subband estimator for modeling subband acoustic transfer function for the time-domain output signal,
a signal path for providing the reference signal in the time-domain in response to a primary signal in the time-domain and the modeling for the time-domain output signal,
a second subband estimator for modeling subband acoustic transfer function for the subband reference signal,
a second analysis filter bank for transferring the primary signal into a plurality of subband primary signals in the frequency-domain,
an adjustor for adjusting the subband adaptive filter in response to the subband primary signals and the modeling for the reference signal, and

a synthesis filter bank for converting the subband adaptive filter to the time-domain filter bank for filtering the reference signal.

10. A system as claimed in any one of claims 1-4, wherein the first over-sampled analysis filterbank is implemented by a Weighted-Overlap Add (WOLA) analysis filterbank, and the over-sampled synthesis filterbank is implemented by a WOLA synthesis filterbank.

11. A system as claimed in any one of claims 1-6 and 10, wherein the subband processing module is provided for implementing one or more than one delayless subband adaptive algorithm.

12. A system as claimed in any one of claims 1-4,10 and 11, wherein the subband processing module is adaptive or re-programmable, and models one or more than one subband transfer function associated with an application of a listening device.

13. A system as claimed in any one of claims 2, 4, 10 and 11, wherein:

the over-sampled synthesis filterbank employs WOLA synthesis process to convert a plurality of subband adaptive filters from the subband processing module into a time-domain filter for filtering the reference signal in the time-domain.

14. A system as claimed in claim 13, wherein a time-domain output signal is provided by the output of the time-domain filter, or a combination of the output of the time-domain filter and the primary signal.

15. A system as claimed in claim 13, wherein:

a time-domain output signal is provided by the output of the time-domain filter and the primary signal, and
the time-domain output signal is provided as input to the second over-sampled analysis filterbank.

16. A system as claimed in any one of claims 3, 4 and 10, wherein the subband processing module is provided for implementing one or more than one subband adaptive algorithm.

17. A system as claimed in any one of claims 1-4 and 10-16, wherein the subband adaptive processing includes Filtered-X LMS (FX-LMS) algorithm.

18. A system as claimed in claim 11, wherein the delayless subband adaptive algorithm includes delayless Filtered-X LMS (FX-LMS) algorithm.

19. A system as claimed in claim 3 or 4, wherein the adjustable parameters of the analog ANC system include loop-filter frequency, loop-gain, similar parameters or combinations thereof.

20. A system as claimed in any one of claims 3-6, wherein the subband processing module is capable of modeling one or more than one acoustic transfer function, a transfer function for a microphone, a transfer function for a loudspeaker, or combinations thereof

21. A system as claimed in claim 7, wherein:

the synthesis filter bank employs WOLA synthesis process to convert the subband adaptive filter into a time-domain filter for filtering the reference signal in the time-domain,

22. A system as claimed in any one of claims 7-9 and 21, wherein the system is implemented by an over-sampled filterbank.

23. A system as claimed in any one of claims 7-9 and 21, wherein the system is implemented by a WOLA filterbank.

24. A system as claimed in any one of claims 7-9, wherein the adjustor implements NMSL adaptation algorithm.

25. A method for implementing active noise cancellation according to any one of claims 1-24.

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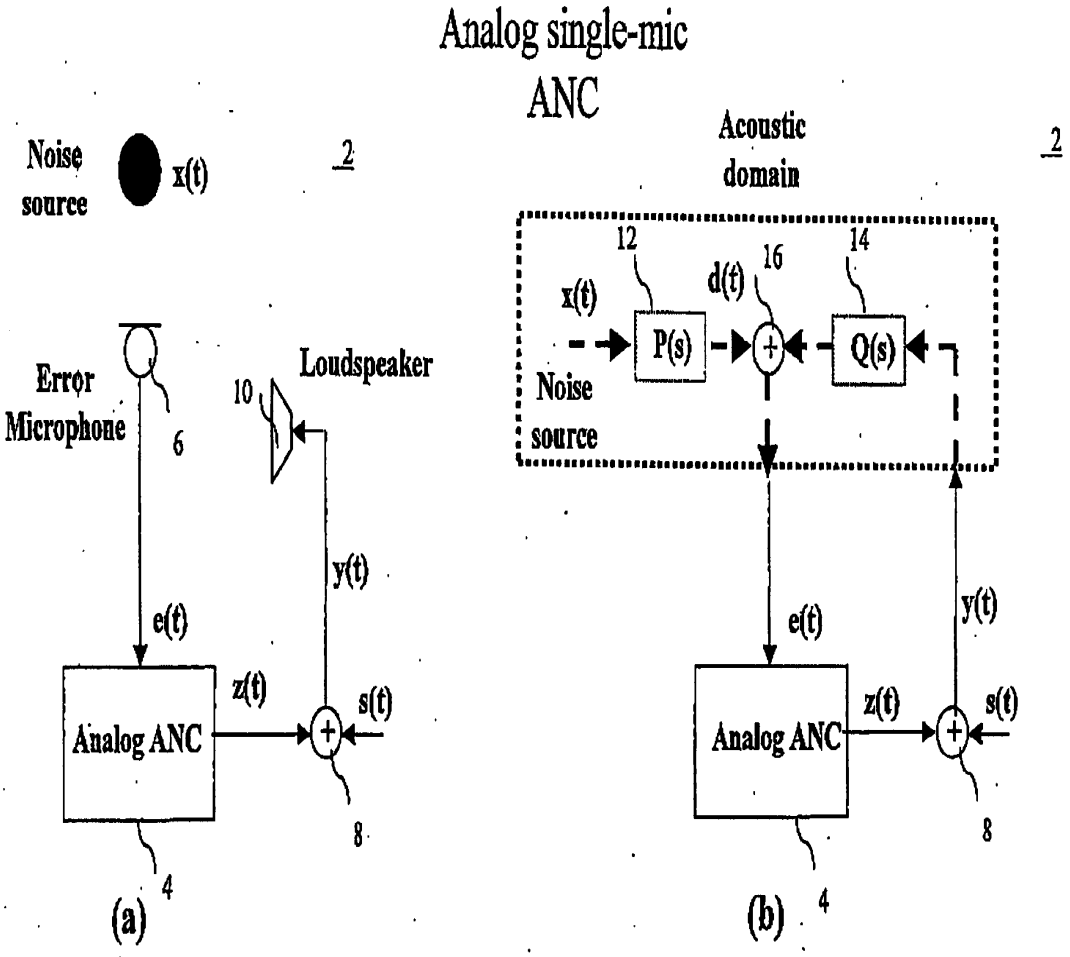


Fig. 1

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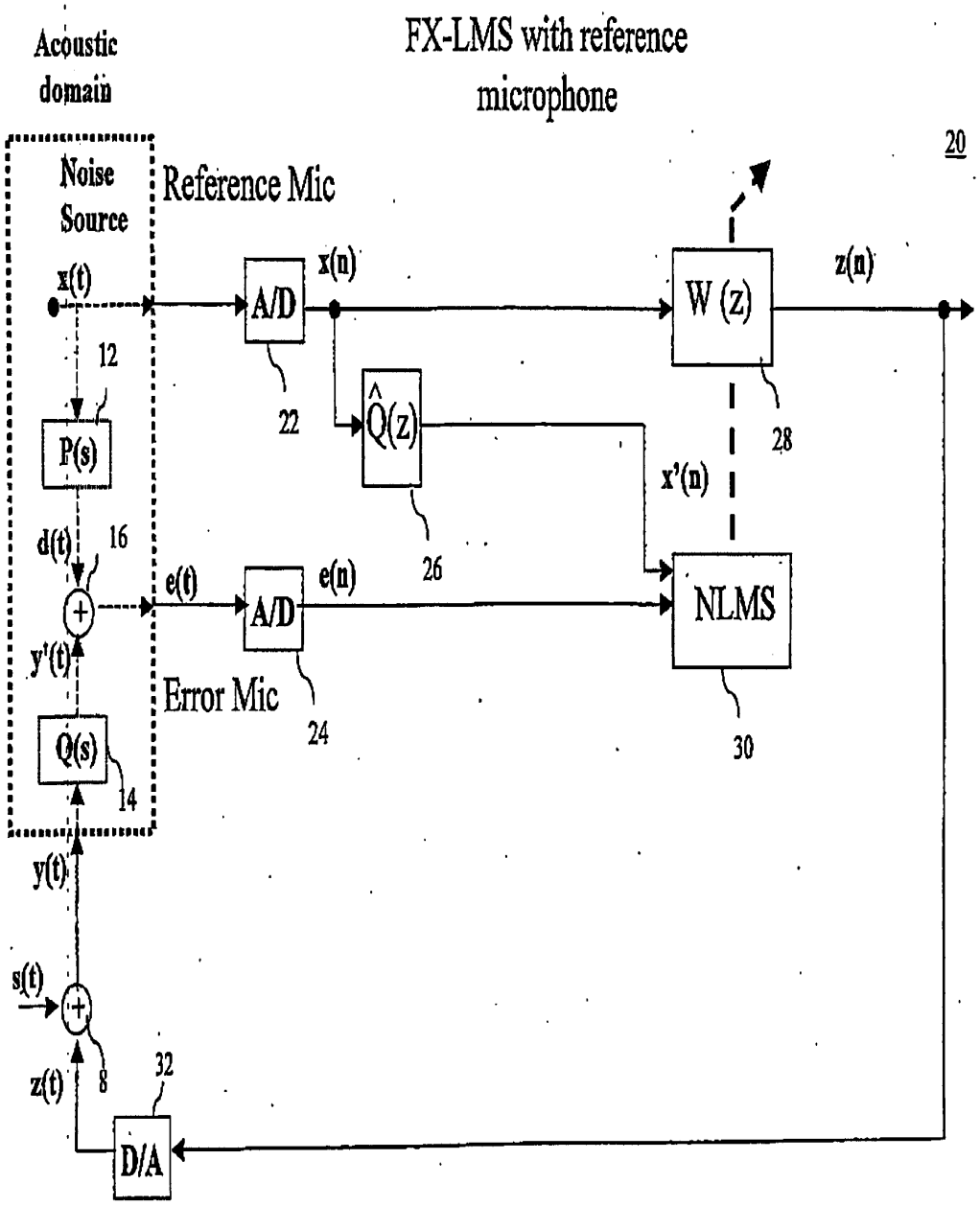

FX-LMS with reference microphone
Acoustic domain
20
Noise Source
Reference Mic
x(t)
A/D
x(n)
W (z)
z(n)
12
22
$\hat{Q}(z)$
28
P(s)
x'(n)
d(t)
16
26
e(t)
A/D
e(n)
NLMS
y'(t)
Error Mic
24
30
Q(s)
14
y(t)
s(t)
z(t)
8
32
D/A

Fig. 2

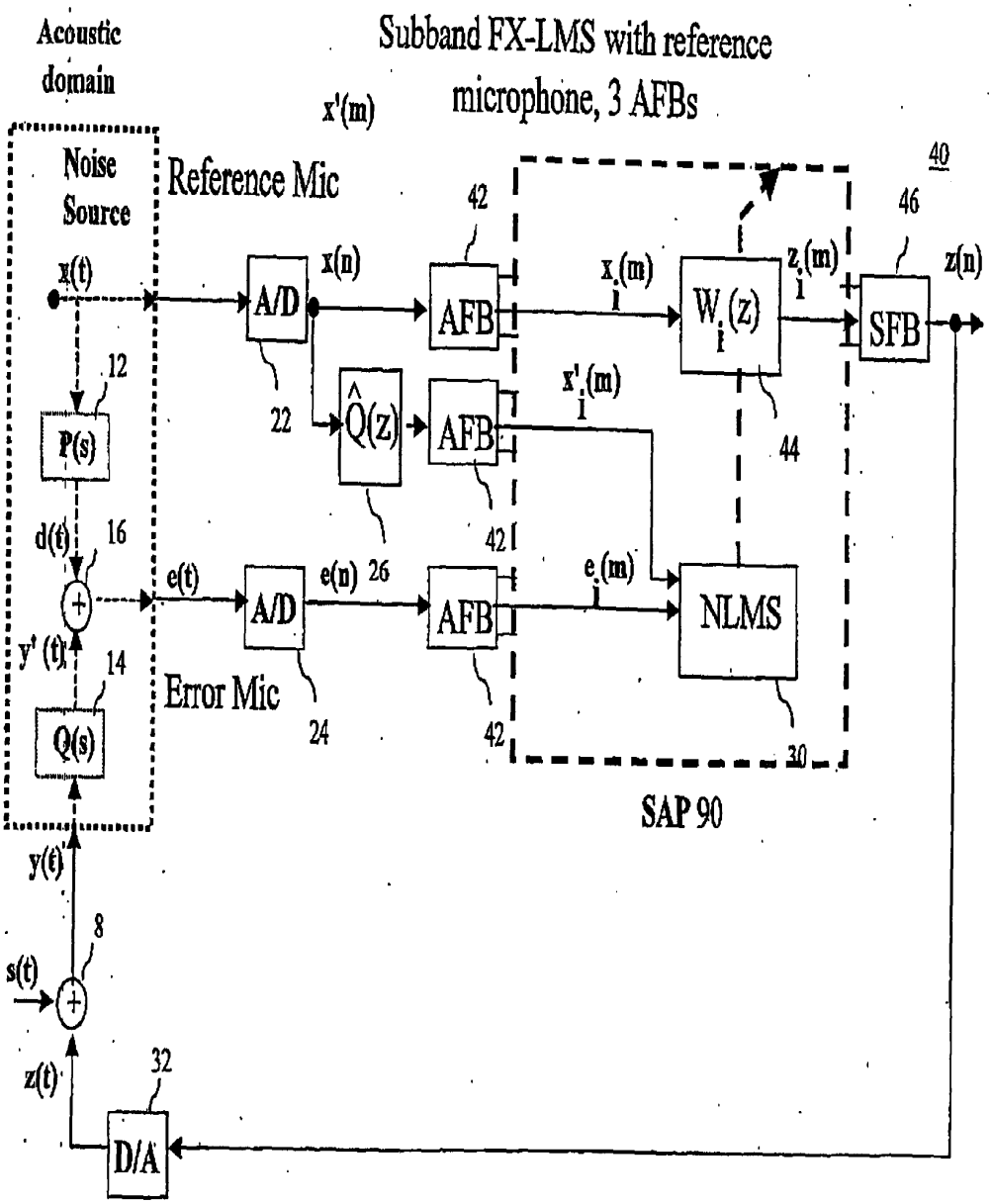


Fig. 3

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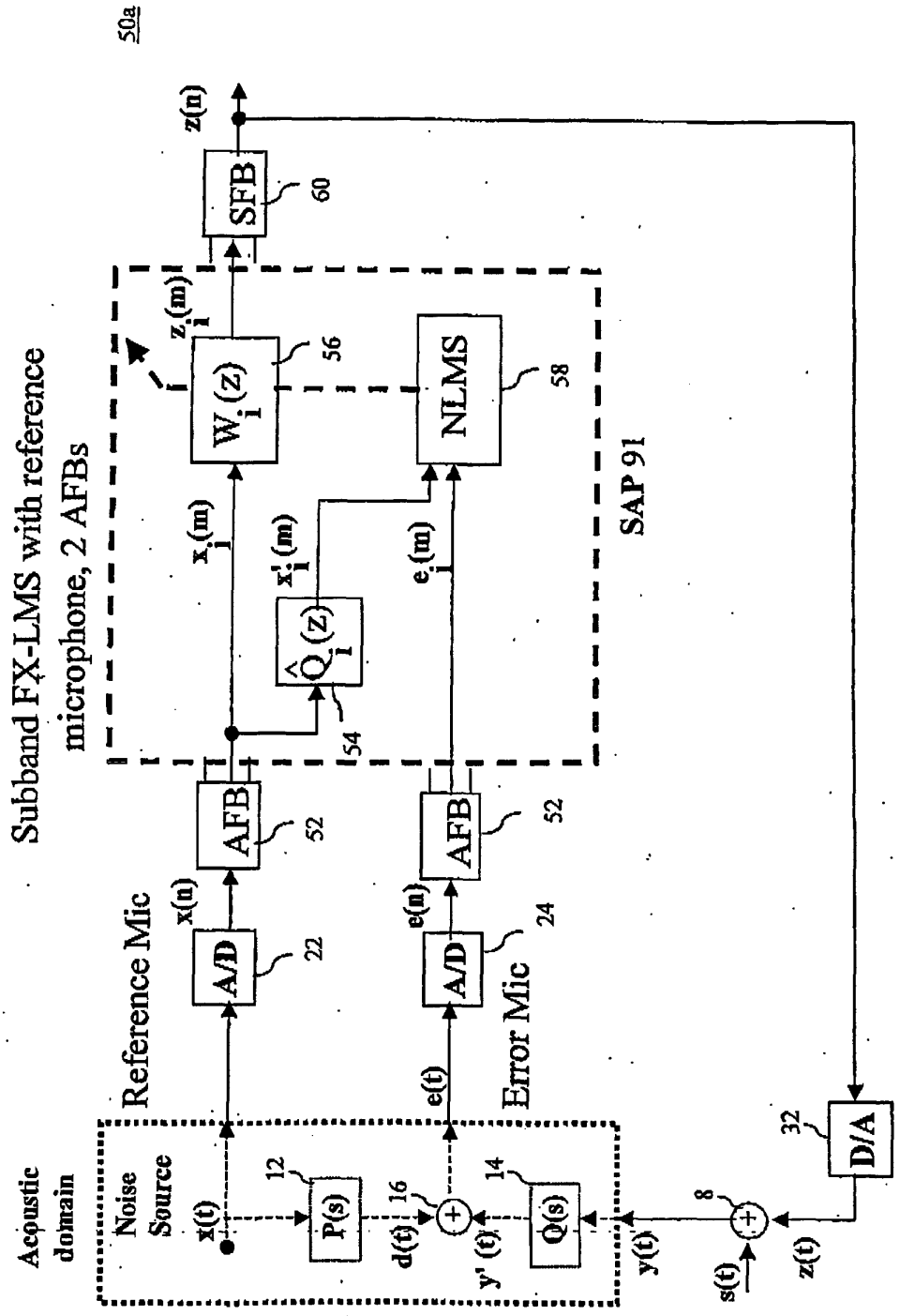


Fig. 4

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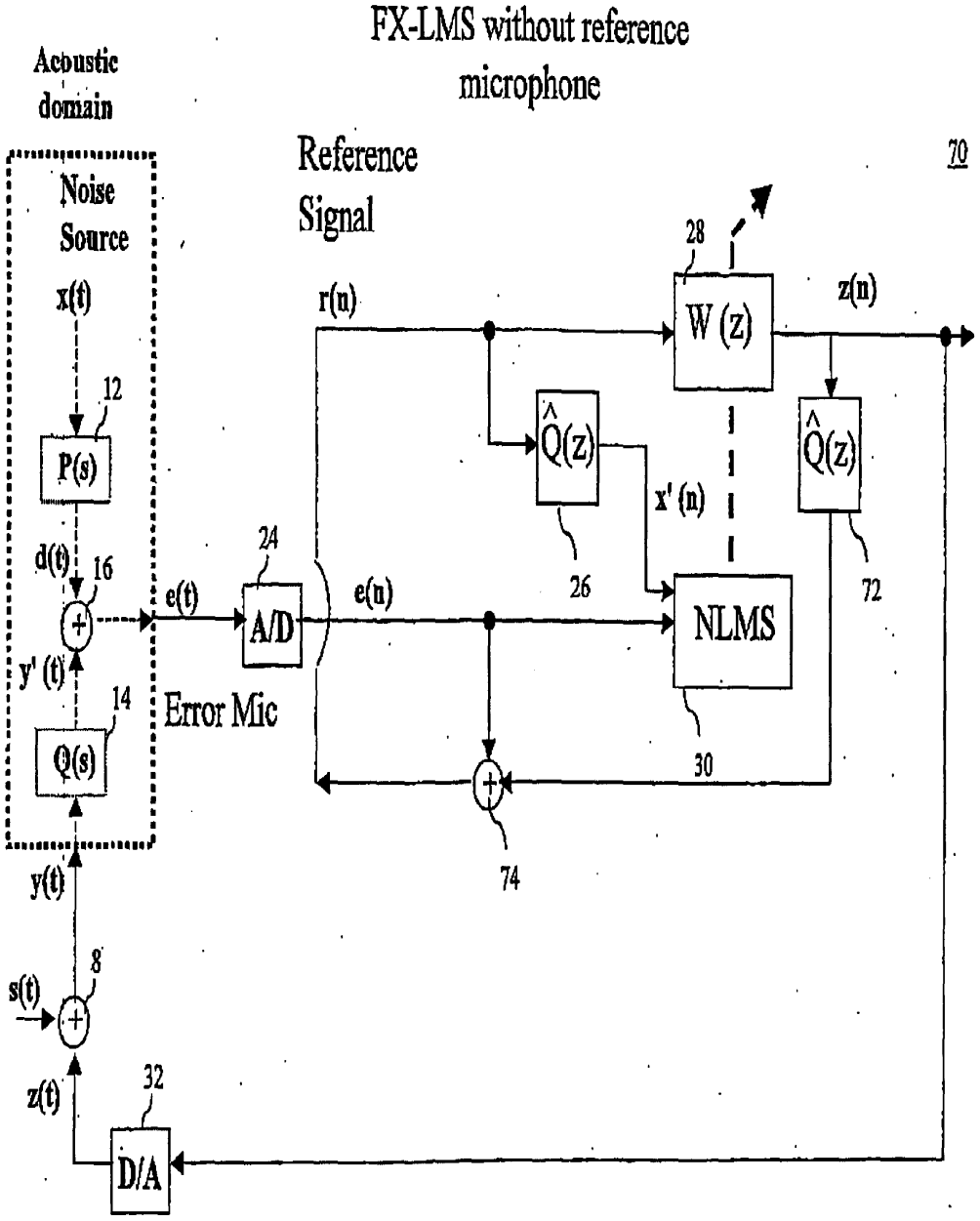

FX-LMS without reference microphone
Acoustic domain
Noise Source
x(t)
12
P(s)
d(t)
16
y' (t)
14
Q(s)
y(t)
s(t)
8
z(t)
32
D/A
Error Mic
e(t)
24
A/D
Reference Signal
r(n)
e(n)
70
28
W (z)
z(n)
Q̂(z)
26
x' (n)
Q̂(z)
72
NLMS
30
74

Fig. 5

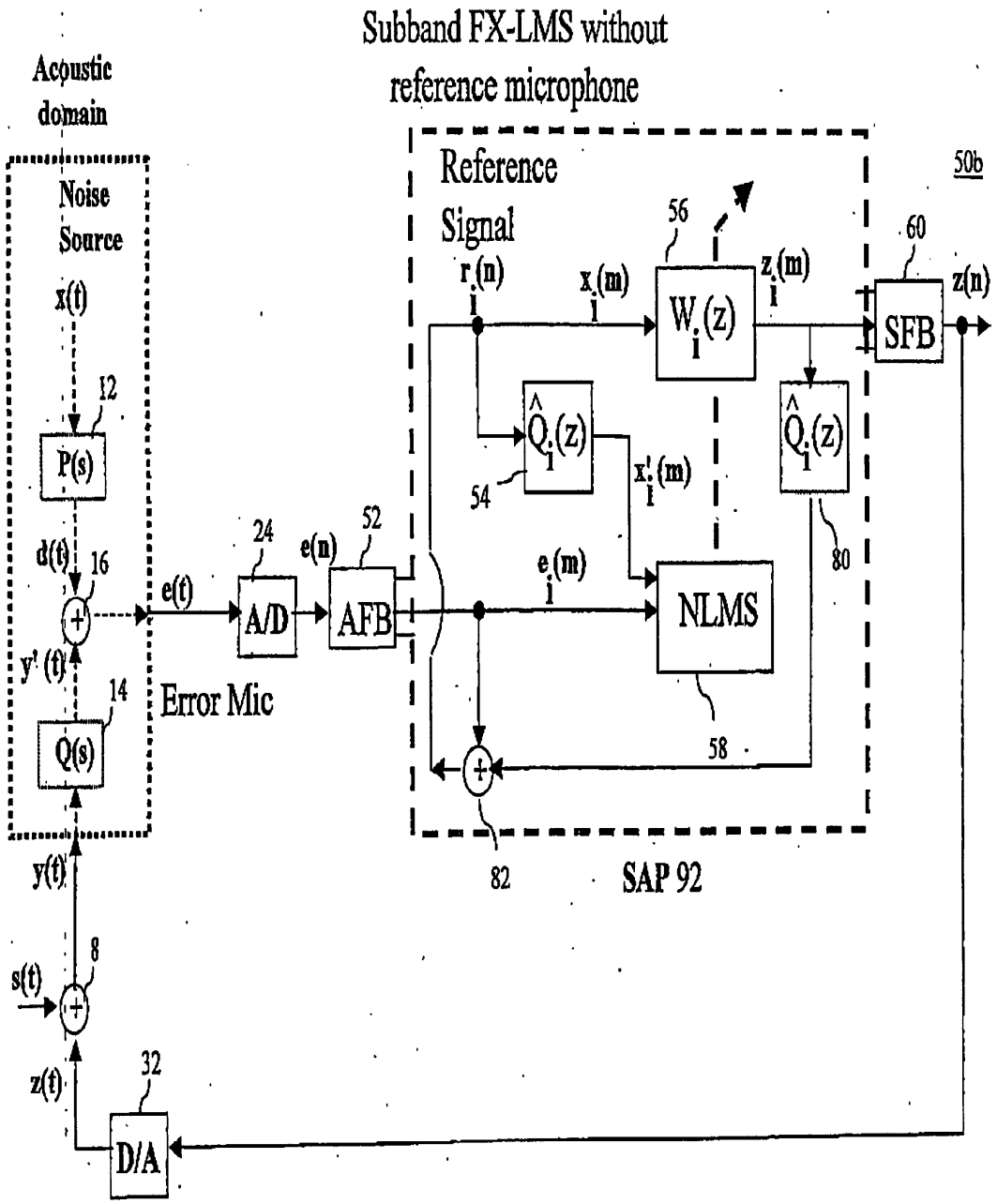

Subband FX-LMS without
reference microphone
Acoustic
domain
Noise
Source
x(t)
12
P(s)
d(t)
16
y' (t)
14
Q(s)
y(t)
8
s(t)
z(t)
32
D/A
e(t)
24
A/D
e(n)
52
AFB
Error Mic
Reference
Signal
r_i(n)
x_i(m)
56
W_i(z)
z_i(m)
60
SFB
z(n)
50b
Q̂_i(z)
54
x'_i(m)
Q̂_i(z)
80
e_i(m)
NLMS
58
82
SAP 92
20
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Fig. 6

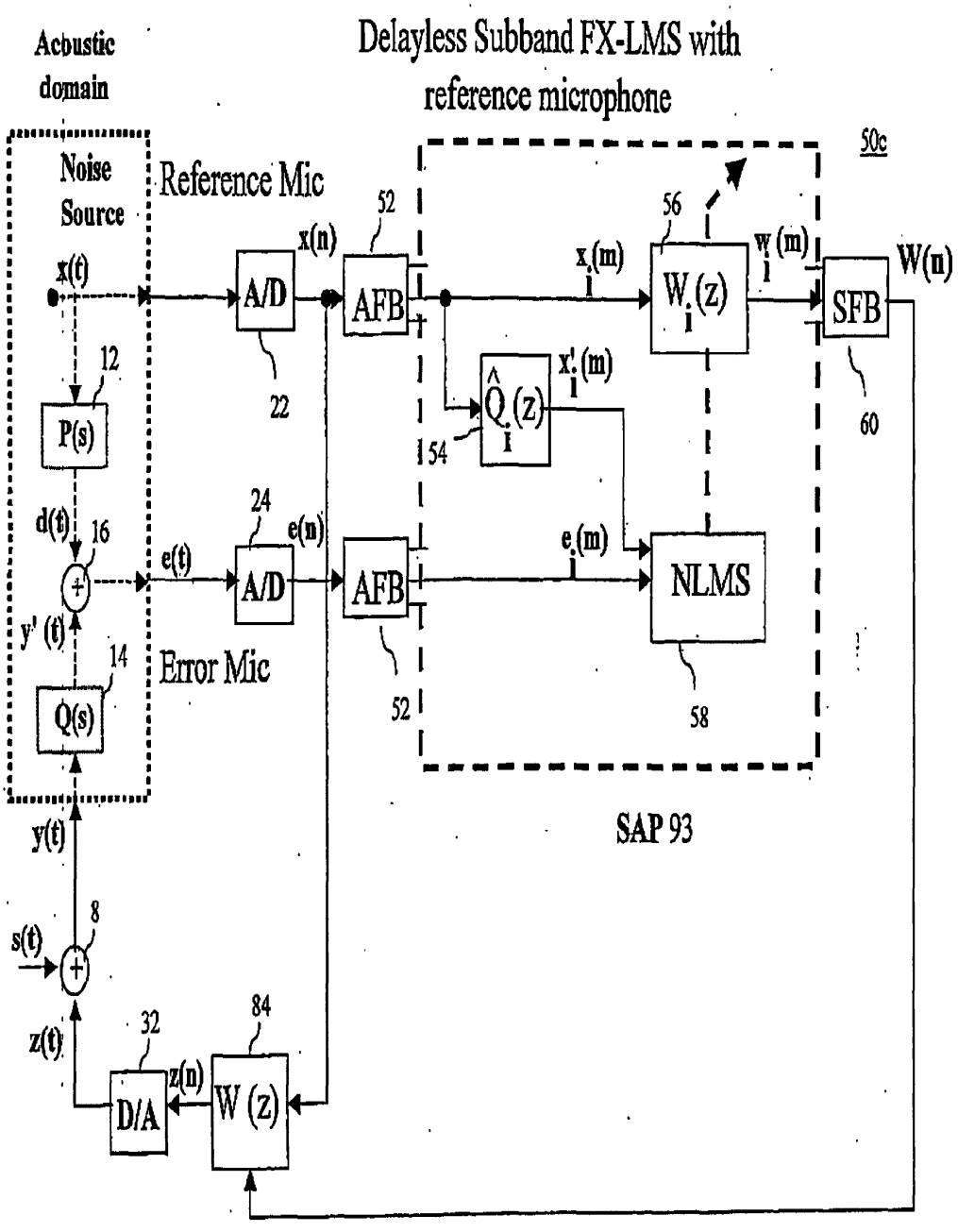


Fig. 7

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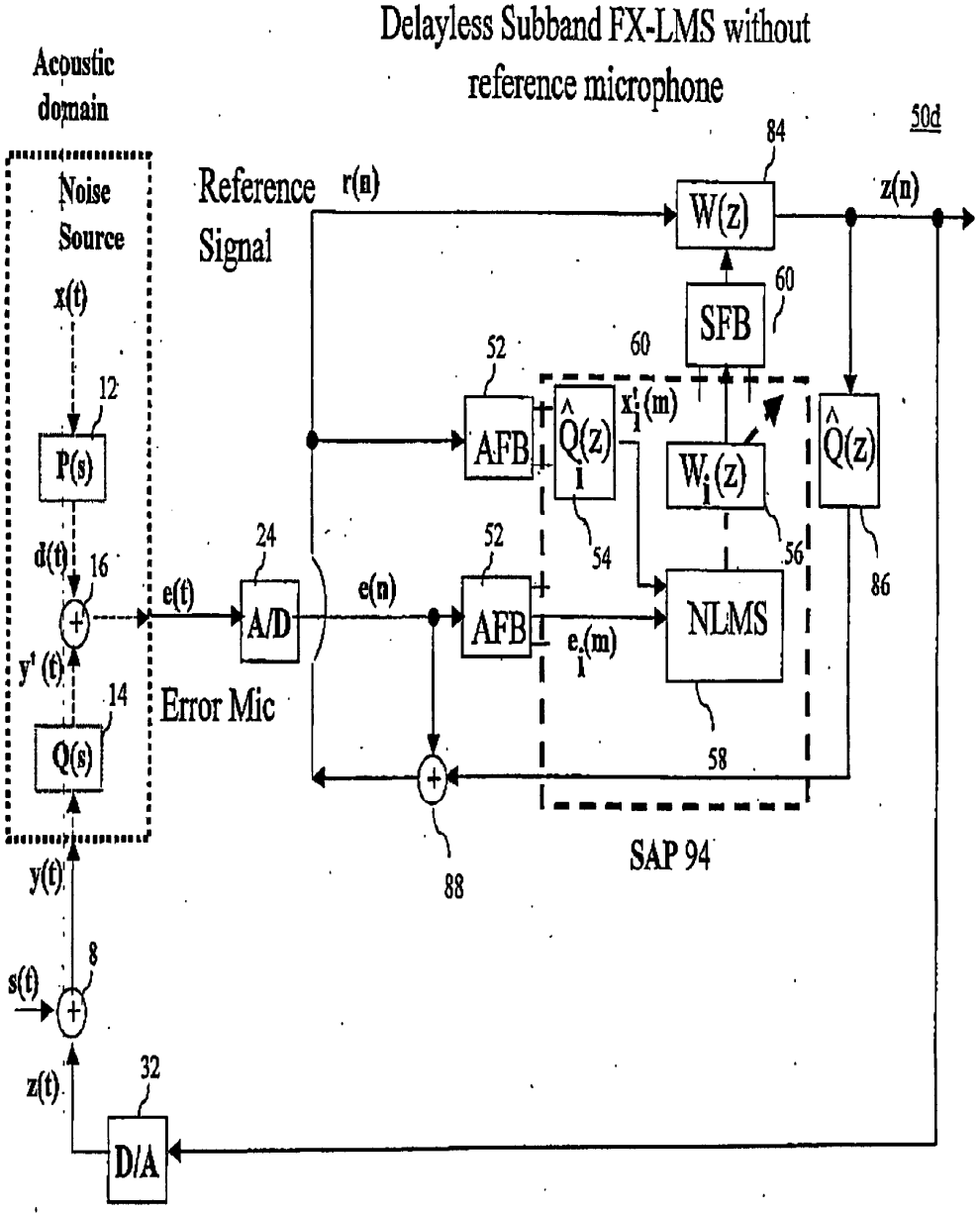


Fig. 8

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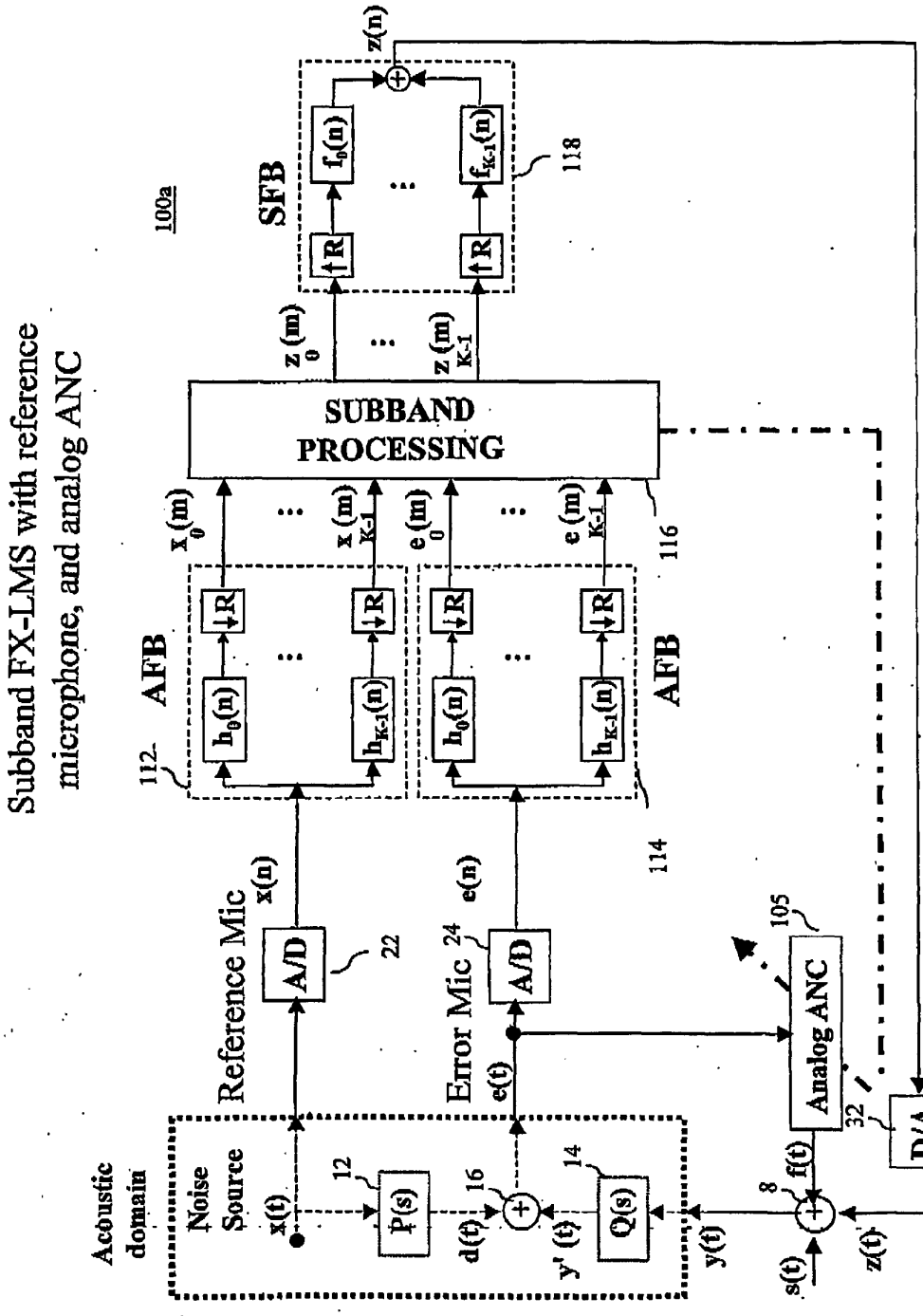

Subband FX-LMS with reference microphone, and analog ANC
100a
SFB
118
SUBBAND PROCESSING
116
AFB
112
AFB
114
Reference Mic
Error Mic
A/D
22
24
Analog ANC
105
32
D/A
Acoustic domain
Noise Source
P(s)
Q(s)
12
14
16
8
x(n)
e(n)
e(t)
d(t)
y'(t)
y(t)
f(t)
s(t)
z(t)
z(n)

Fig. 9

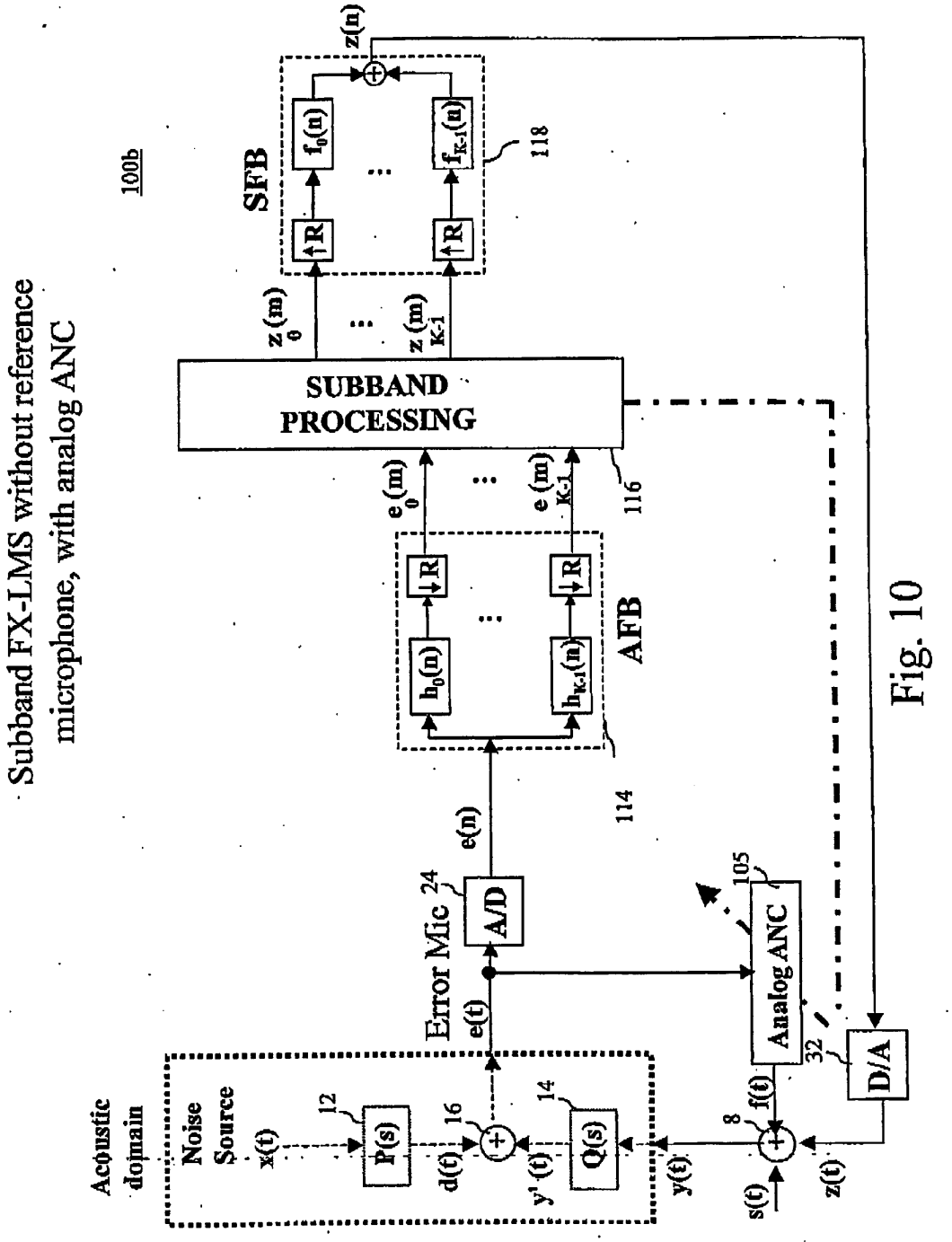

Subband FX-LMS without reference microphone, with analog ANC
100b
Acoustic domain
Noise Source
x(t)
12
P(s)
d(t)
16
y'(t)
14
Q(s)
y(t)
Error Mic
e(t)
24
A/D
e(n)
114
AFB
$h_0(n)$
$h_{K-1}(n)$
R
$e(m)_0$
$e(m)_{K-1}$
116
SUBBAND PROCESSING
$z(m)_0$
$z(m)_{K-1}$
SFB
$f_0(n)$
$f_{K-1}(n)$
118
z(n)
105
Analog ANC
32
D/A
f(t)
8
s(t)
z(t)

Fig. 10

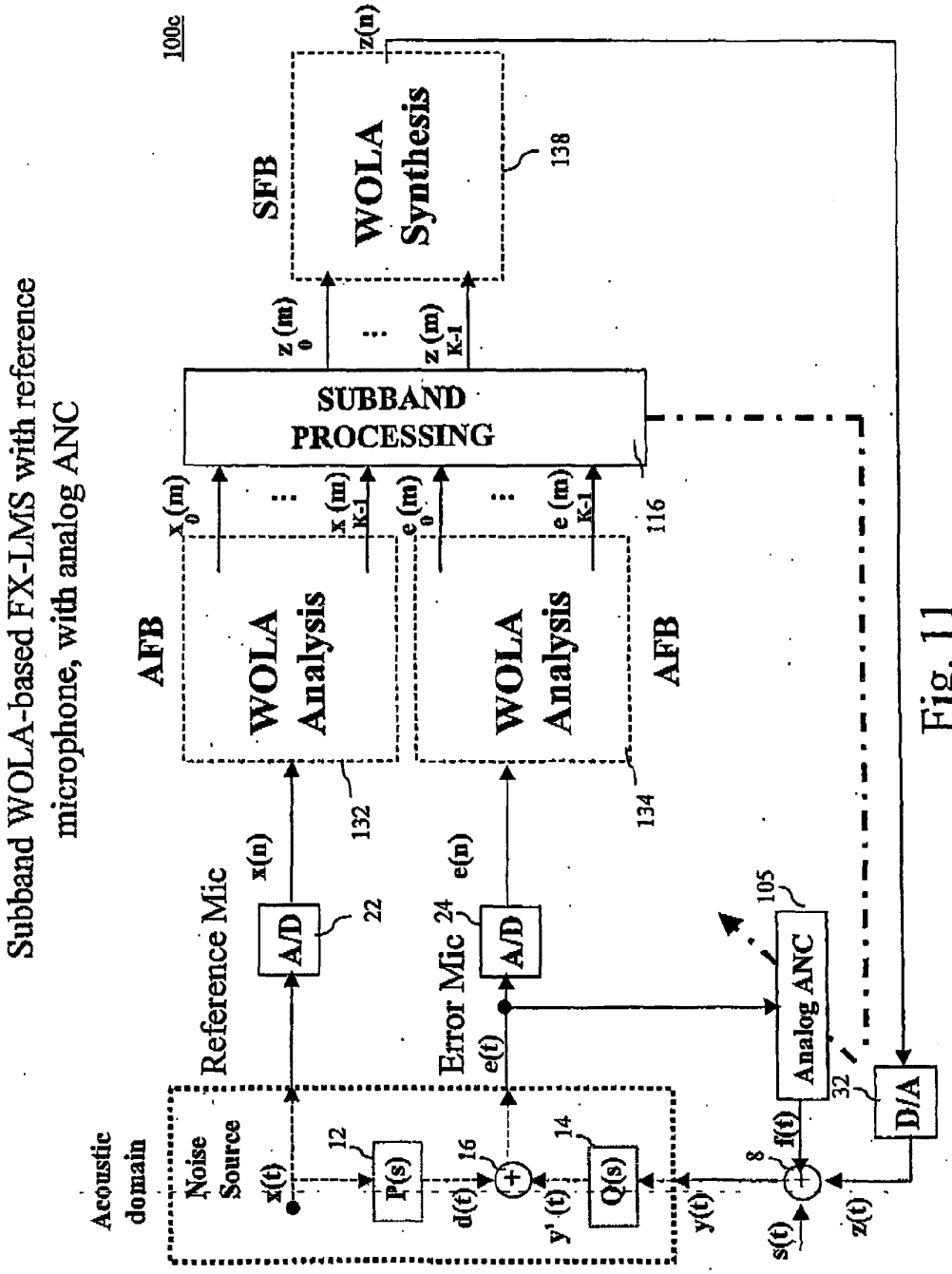


Fig. 11

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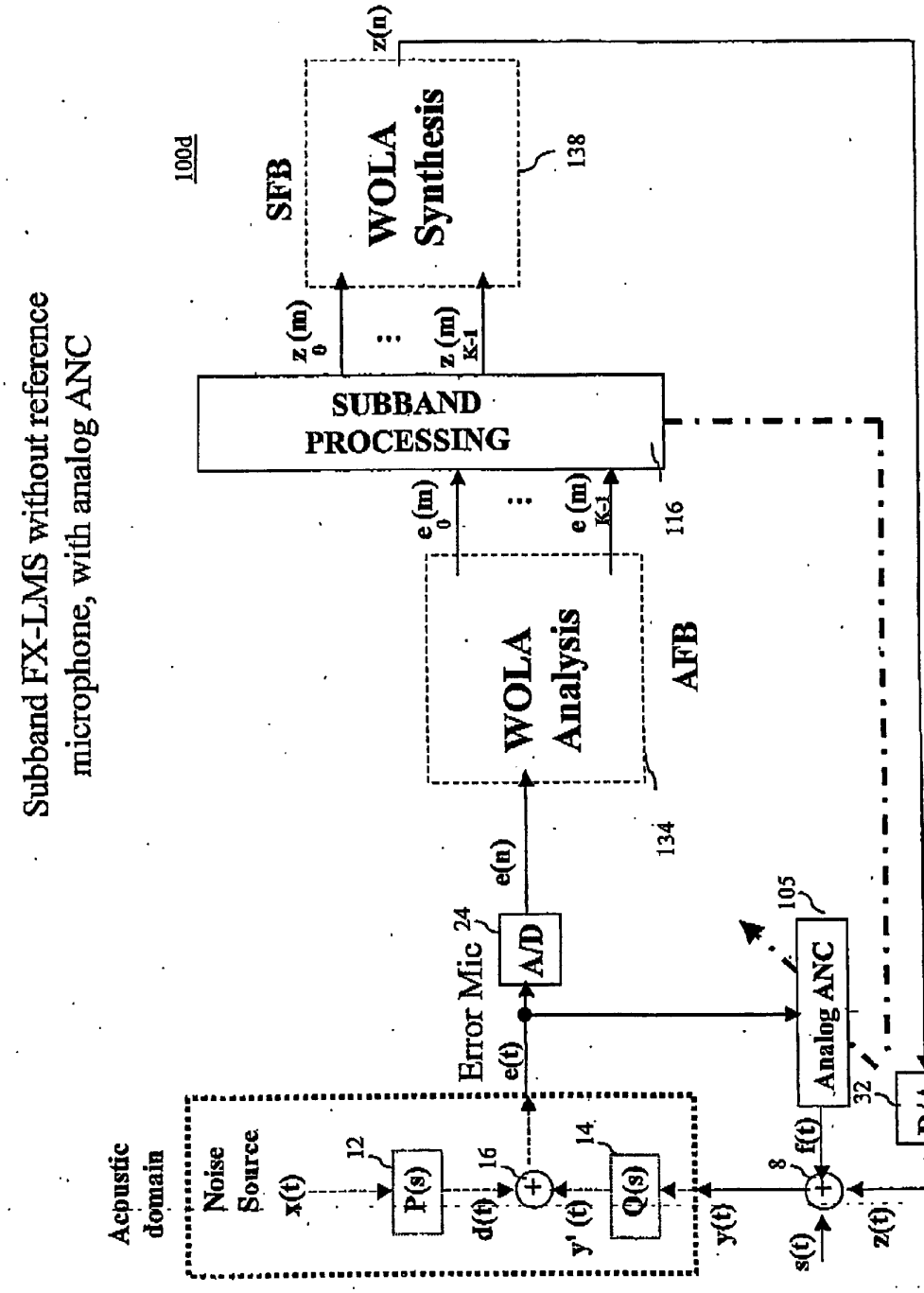


Fig. 12

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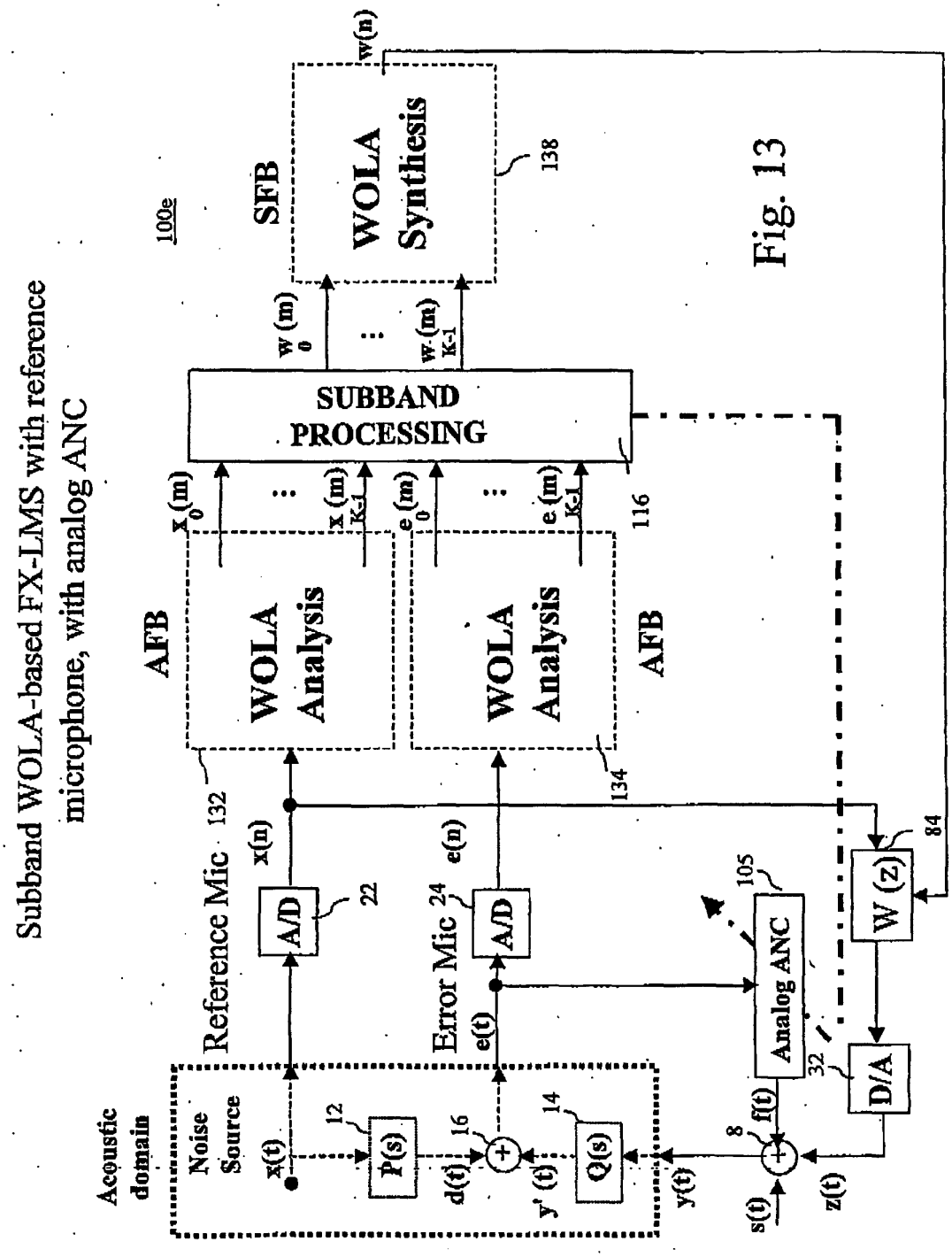


Fig. 13

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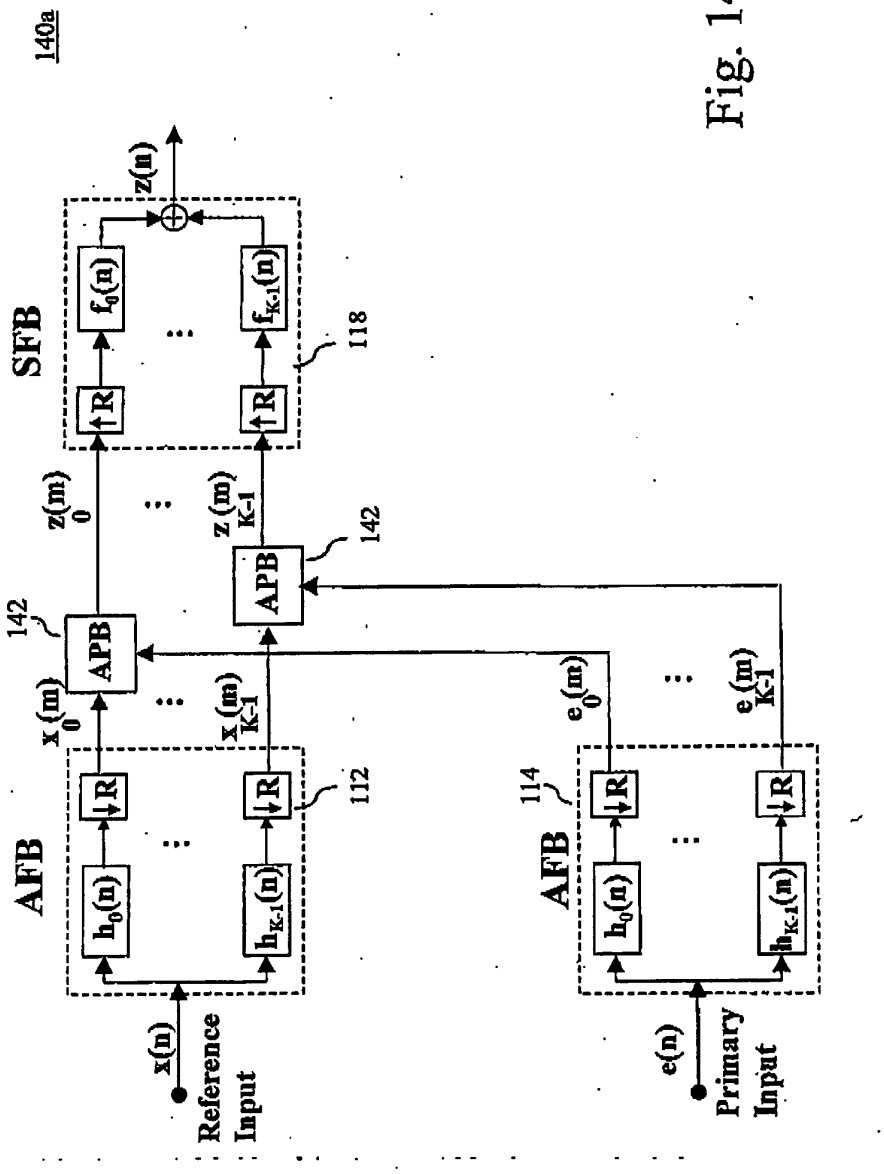

140a
SFB
AFB
AFB
APB
APB
APB
142
142
142
118
112
114
z(n)
x(n)
Reference Input
e(n)
Primary Input
R
$f_0(n)$
$f_{K-1}(n)$
$h_0(n)$
$h_{K-1}(n)$
$z(m)_0$
$z(m)_{K-1}$
$x(m)_0$
$x(m)_{K-1}$
$e(m)_0$
$e(m)_{K-1}$

Fig. 14

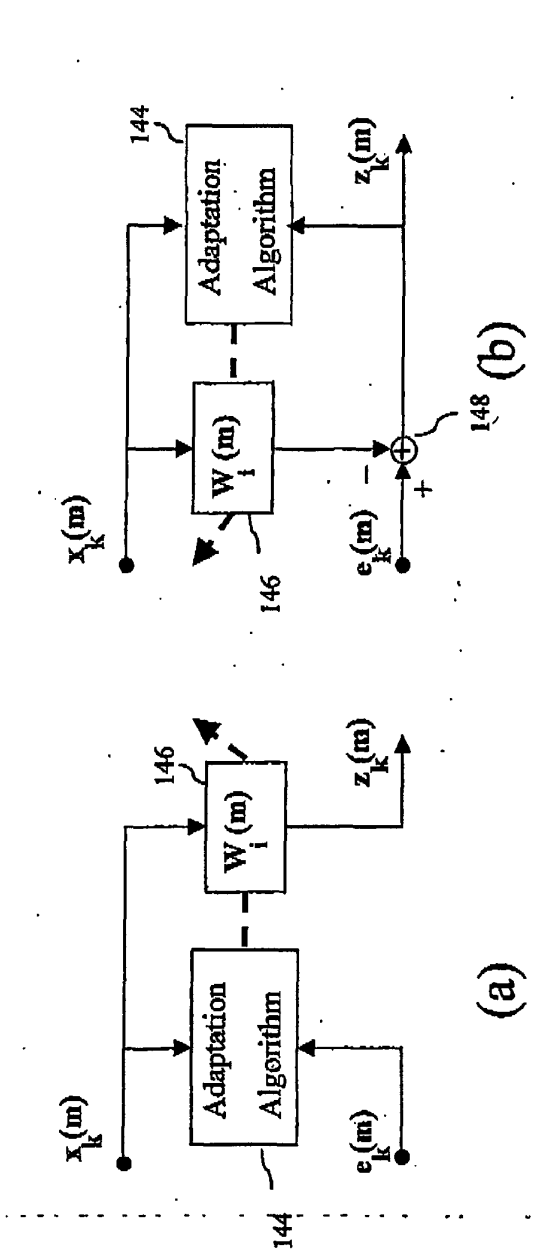


Fig. 15

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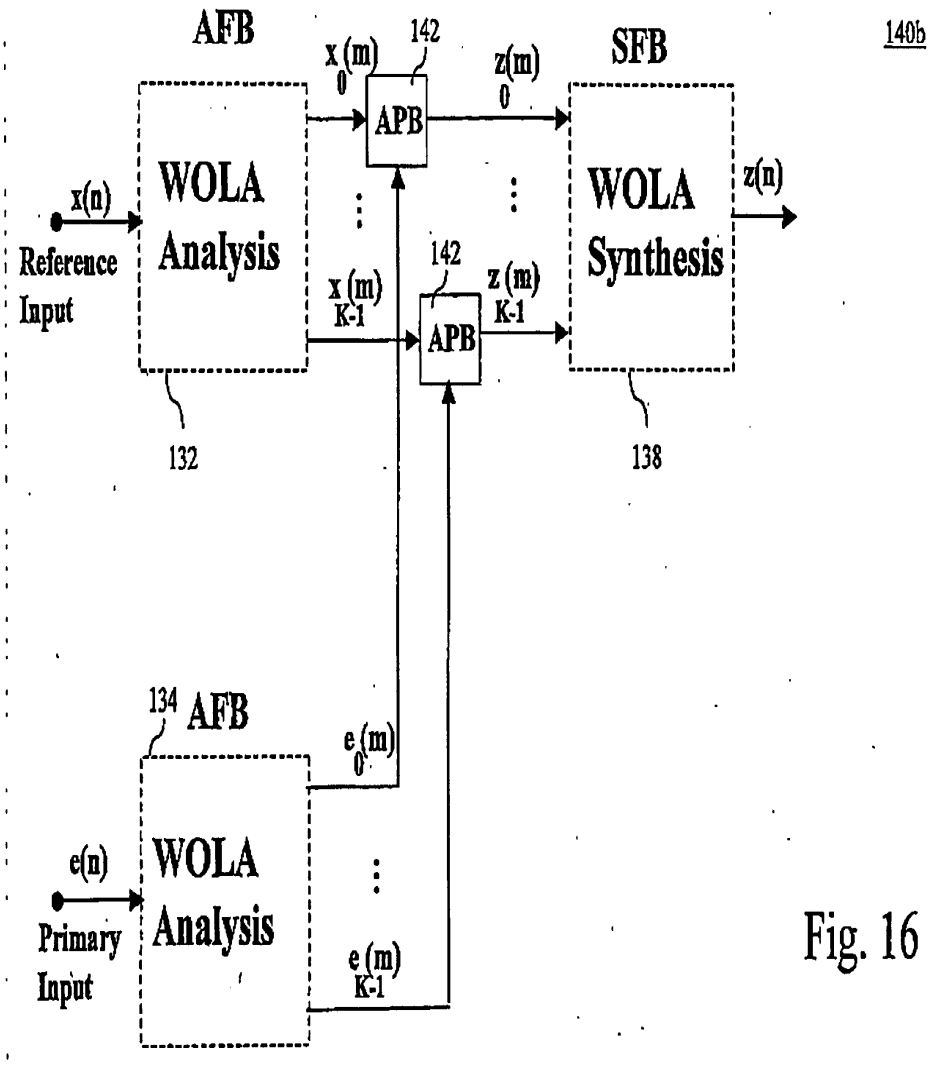

AFB
x(m) 0
142
APB
z(m) 0
SFB
140b
x(n)
Reference Input
WOLA Analysis
WOLA Synthesis
z(n)
x(m) K-1
142
APB
z(m) K-1
132
138
134 AFB
e(m) 0
e(n)
Primary Input
WOLA Analysis
e(m) K-1

Fig. 16

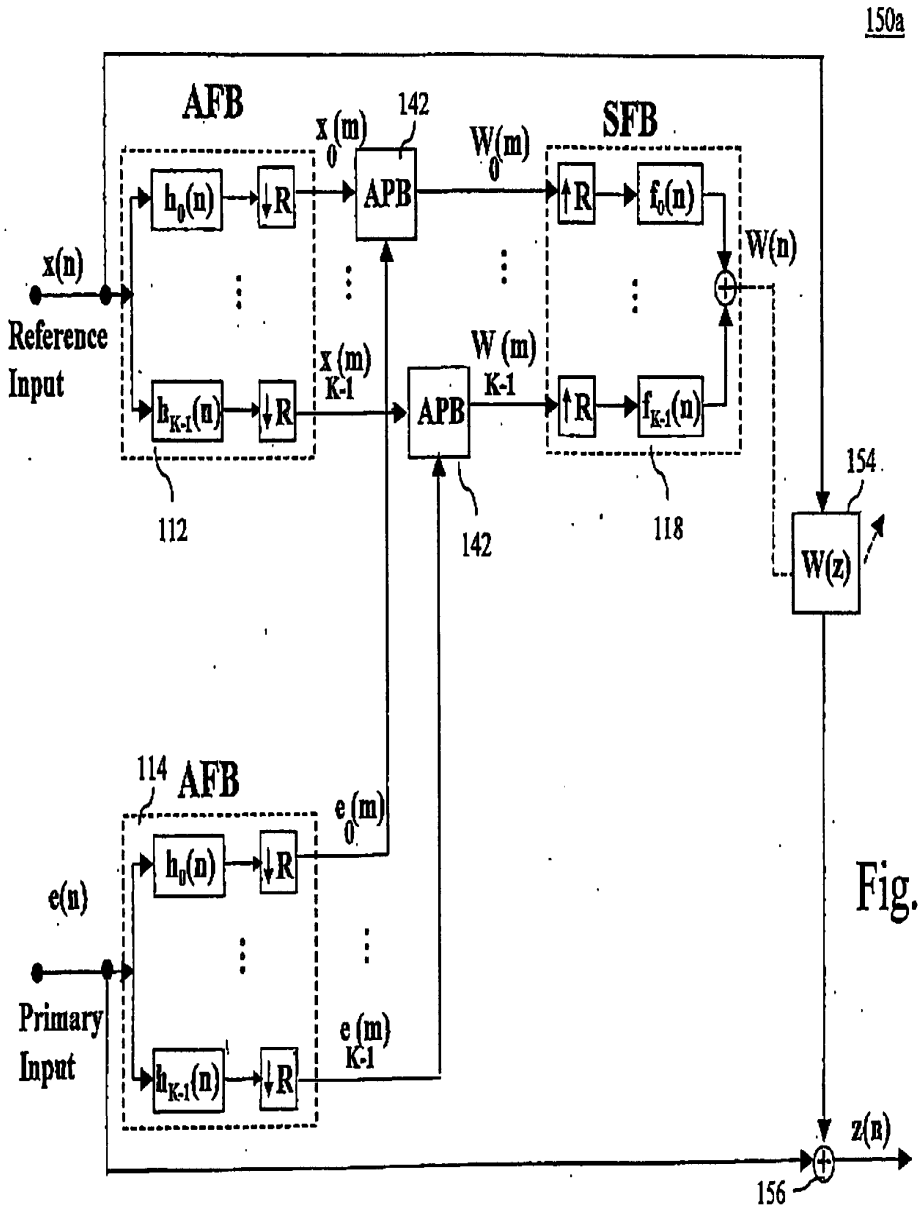

150a
AFB
SFB
142
APB
APB
x(n)
Reference
Input
W(n)
154
W(z)
112
118
142
114
AFB
e(n)
Primary
Input
z(n)
156

Fig. 17

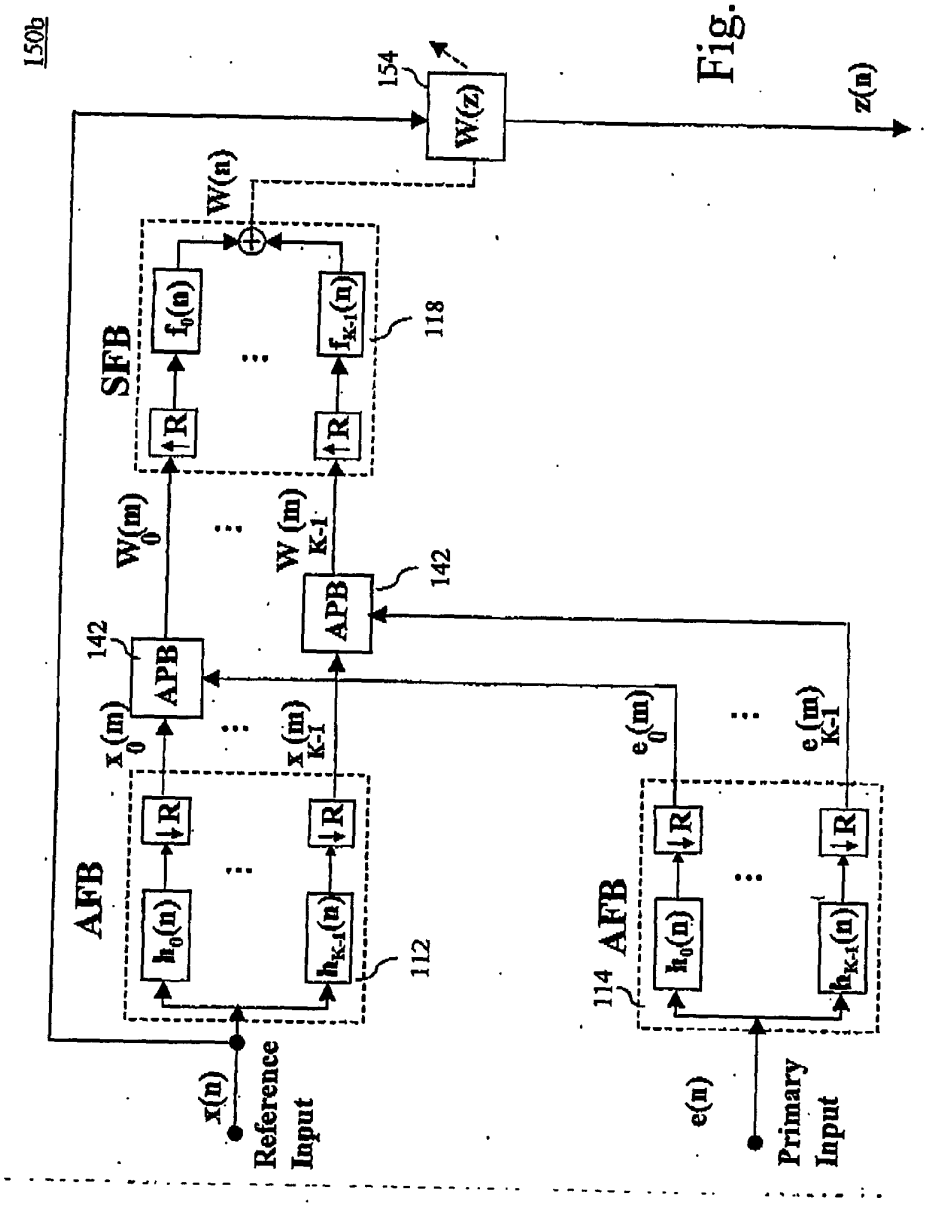


Fig. 18

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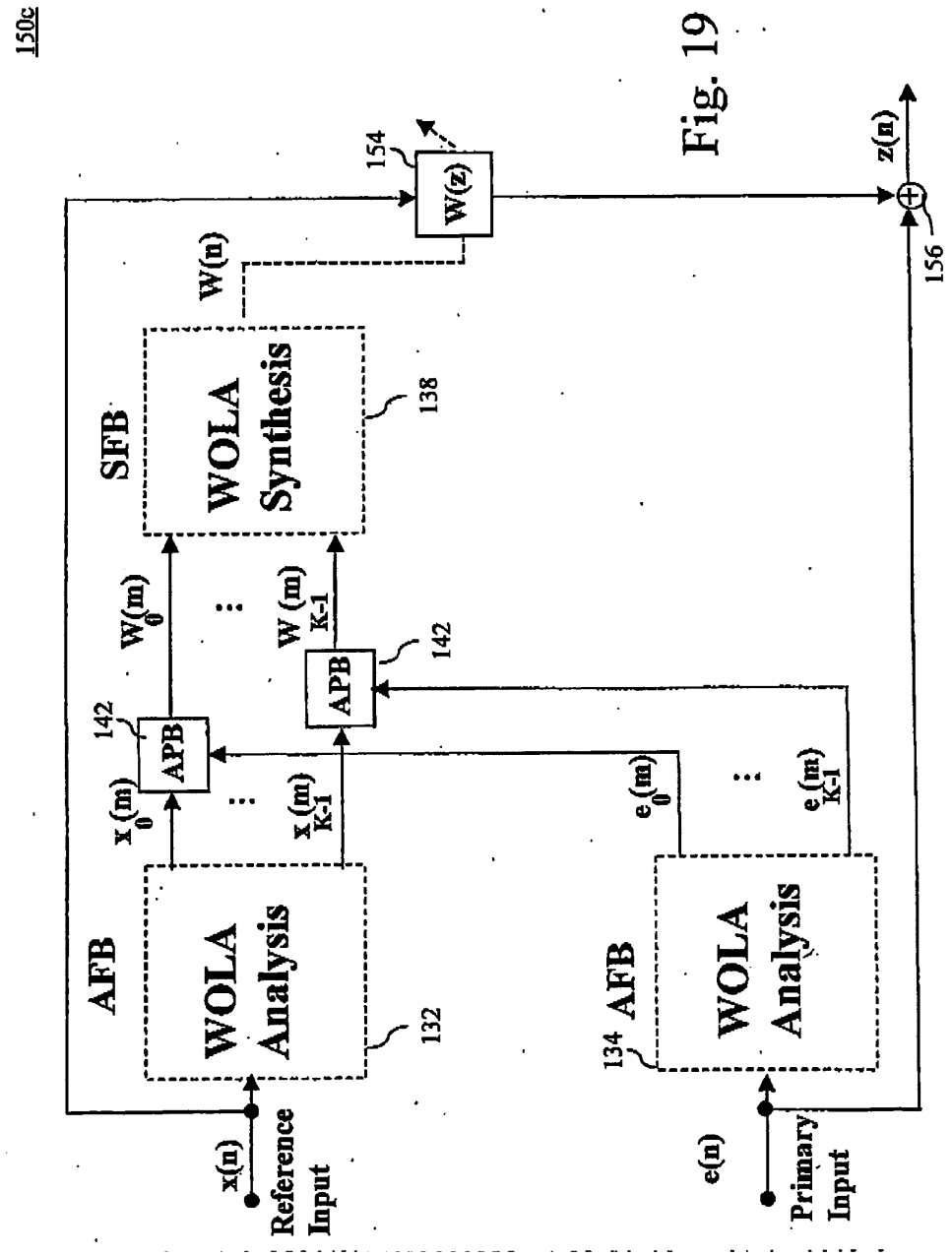

150c
154
W(z)
z(n)
156
W(n)
SFB
WOLA Synthesis
138
$W_0(m)$
$W_{K-1}(m)$
142
APB
142
APB
$x_0(m)$
$x_{K-1}(m)$
AFB
WOLA Analysis
132
$e_0(m)$
$e_{K-1}(m)$
134 AFB
WOLA Analysis
x(n)
Reference Input
e(n)
Primary Input

Fig. 19

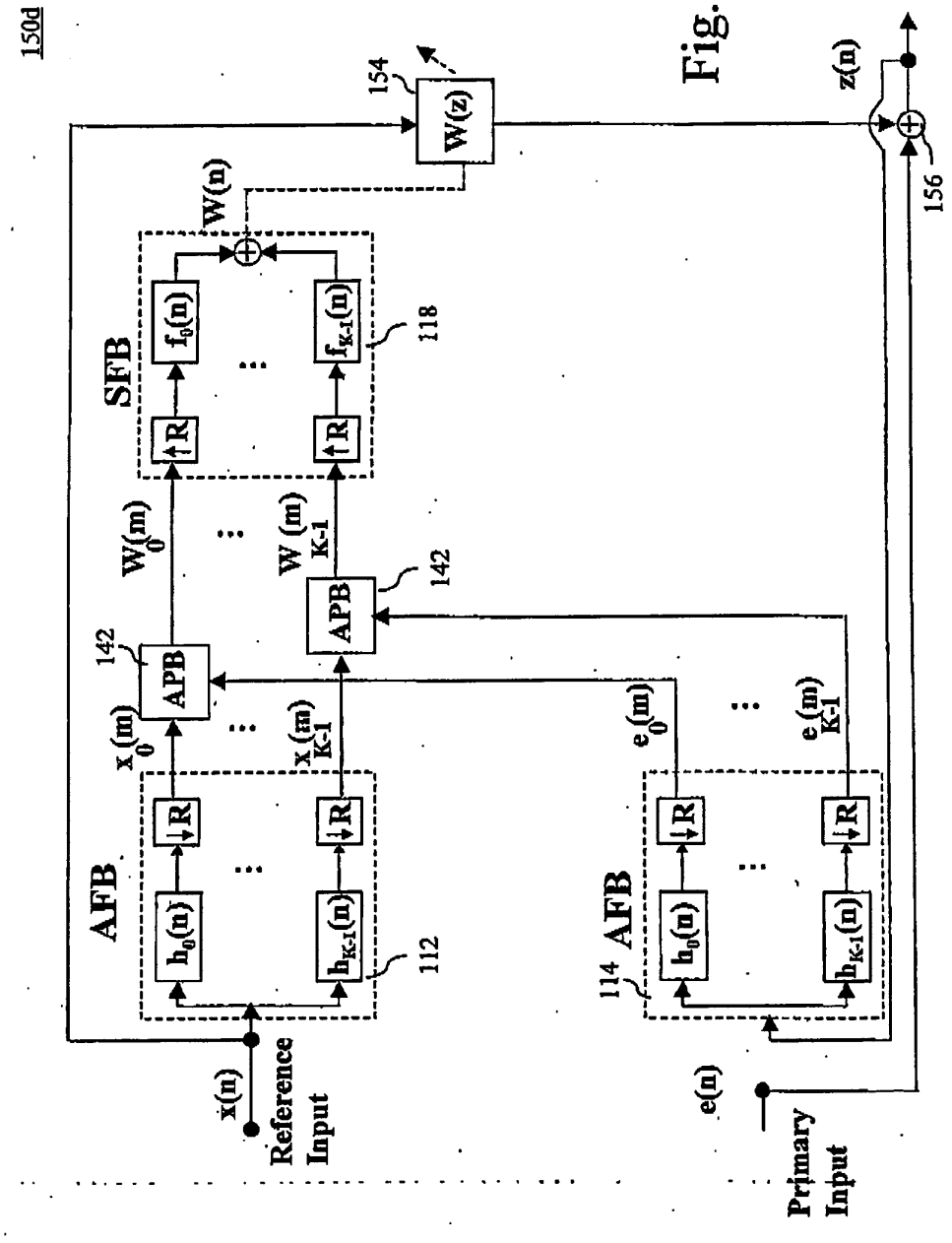

150d
154
W(z)
z(n)
156
W(n)
SFB
118
$f_0(n)$
$f_{K-1}(n)$
R
$W_0(m)$
$W_{K-1}(m)$
142
APB
$x_0(m)$
$x_{K-1}(m)$
AFB
112
$h_0(n)$
$h_{K-1}(n)$
x(n)
Reference Input
114
$e_0(m)$
$e_{K-1}(m)$
e(n)
Primary Input

Fig. 20

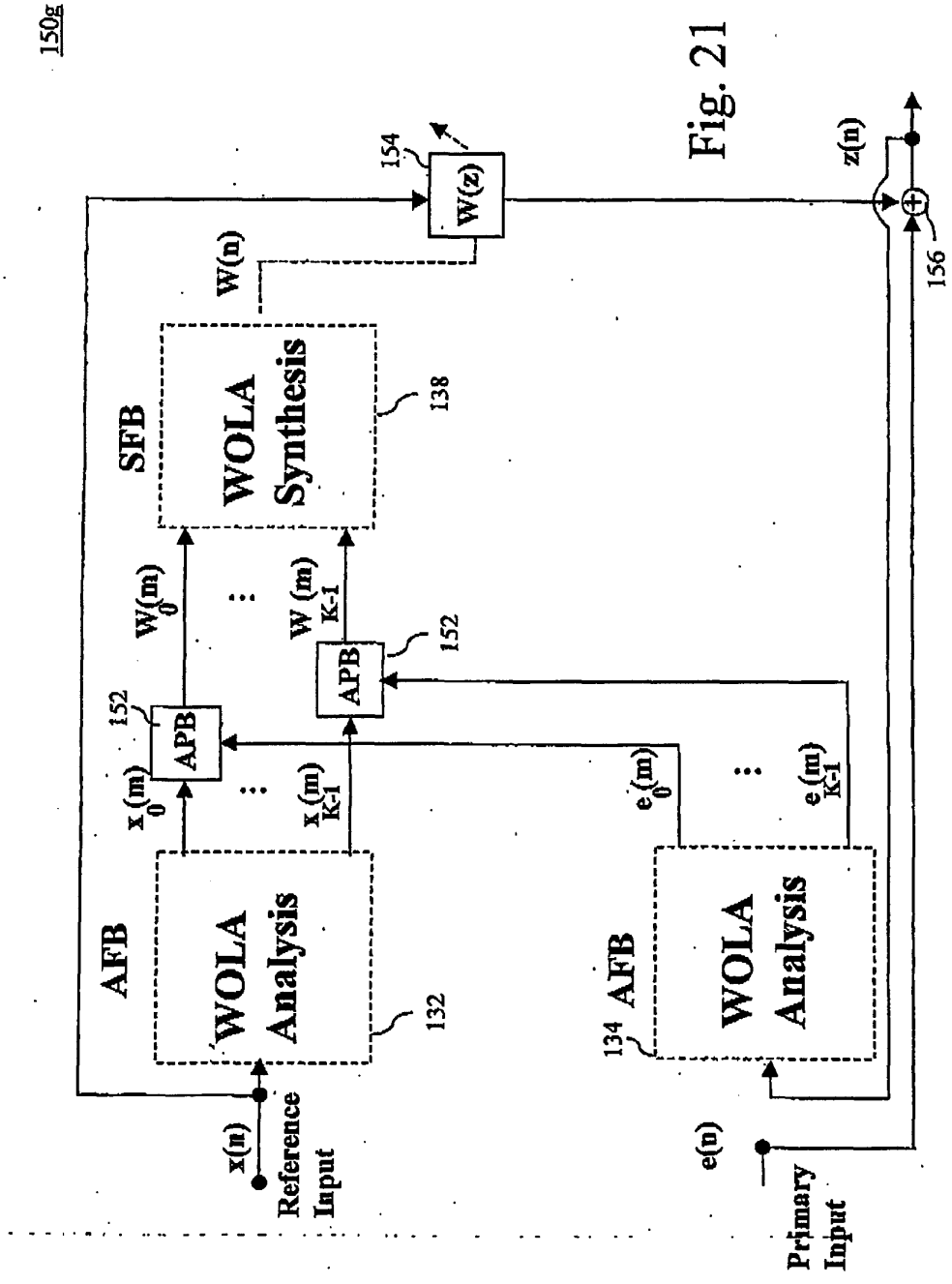


Fig. 21

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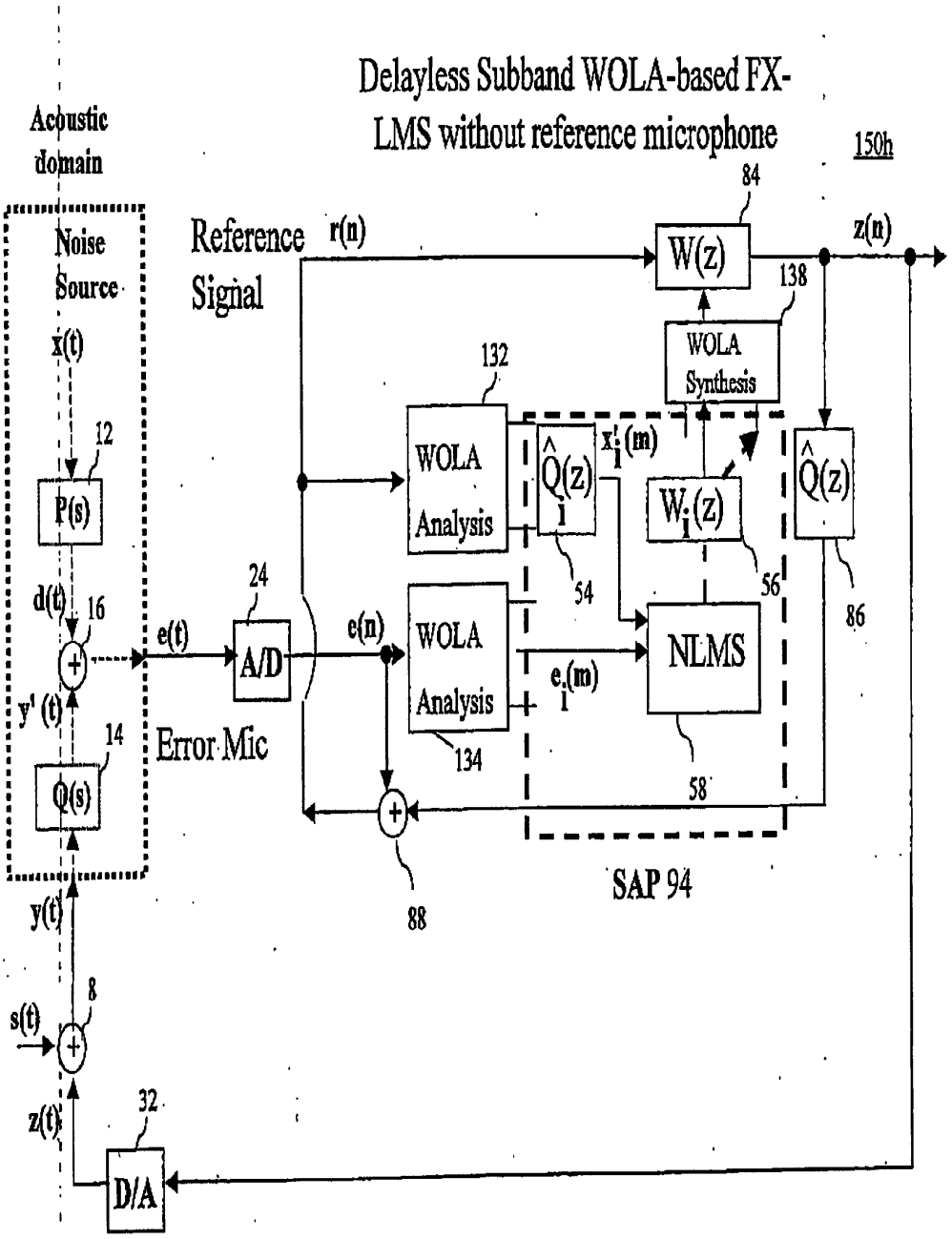


Fig. 22

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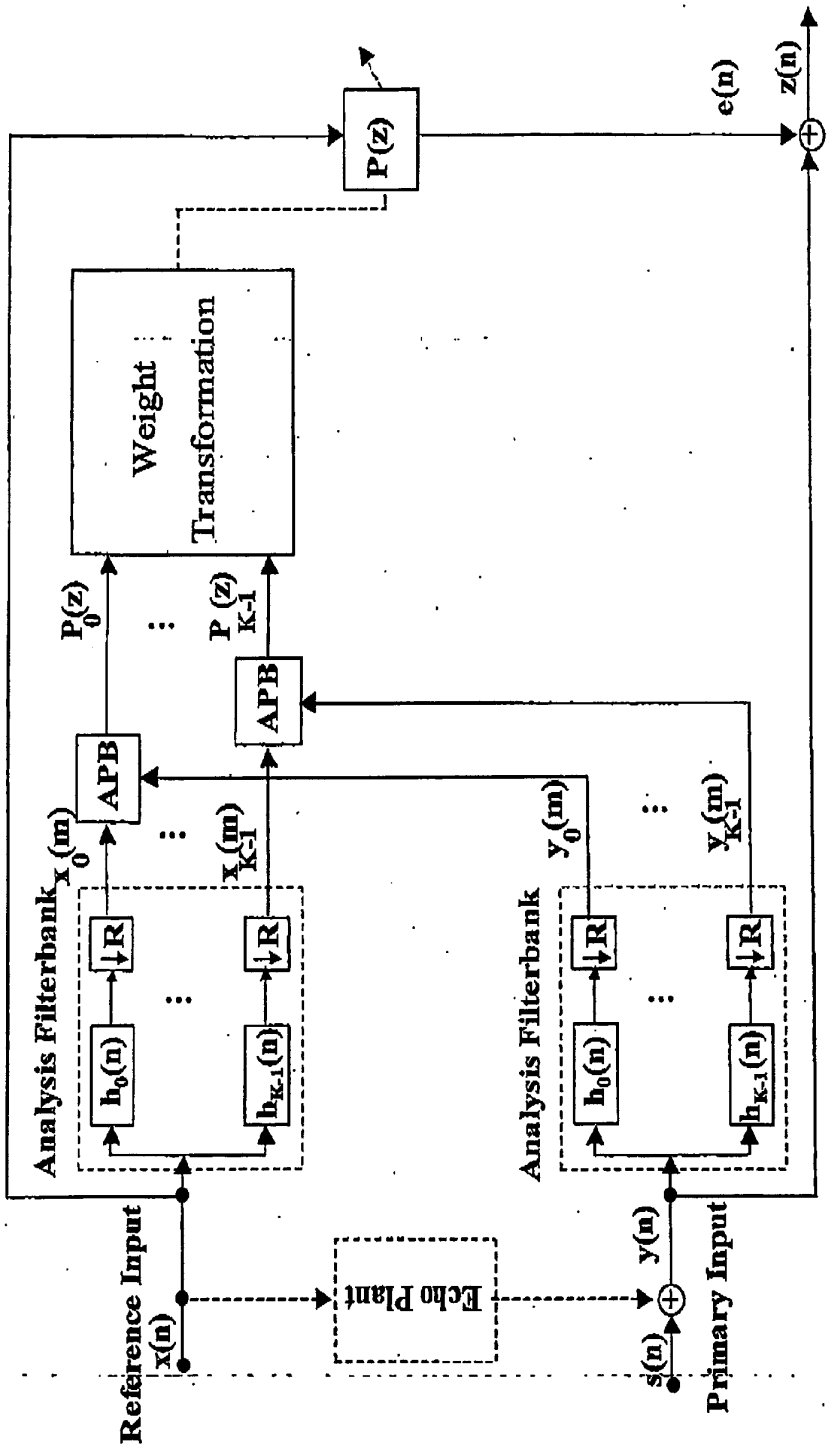


FIG.23: Delayless SAF system using oversampled filterbanks, employing weight transform for time-filter reconstruction.

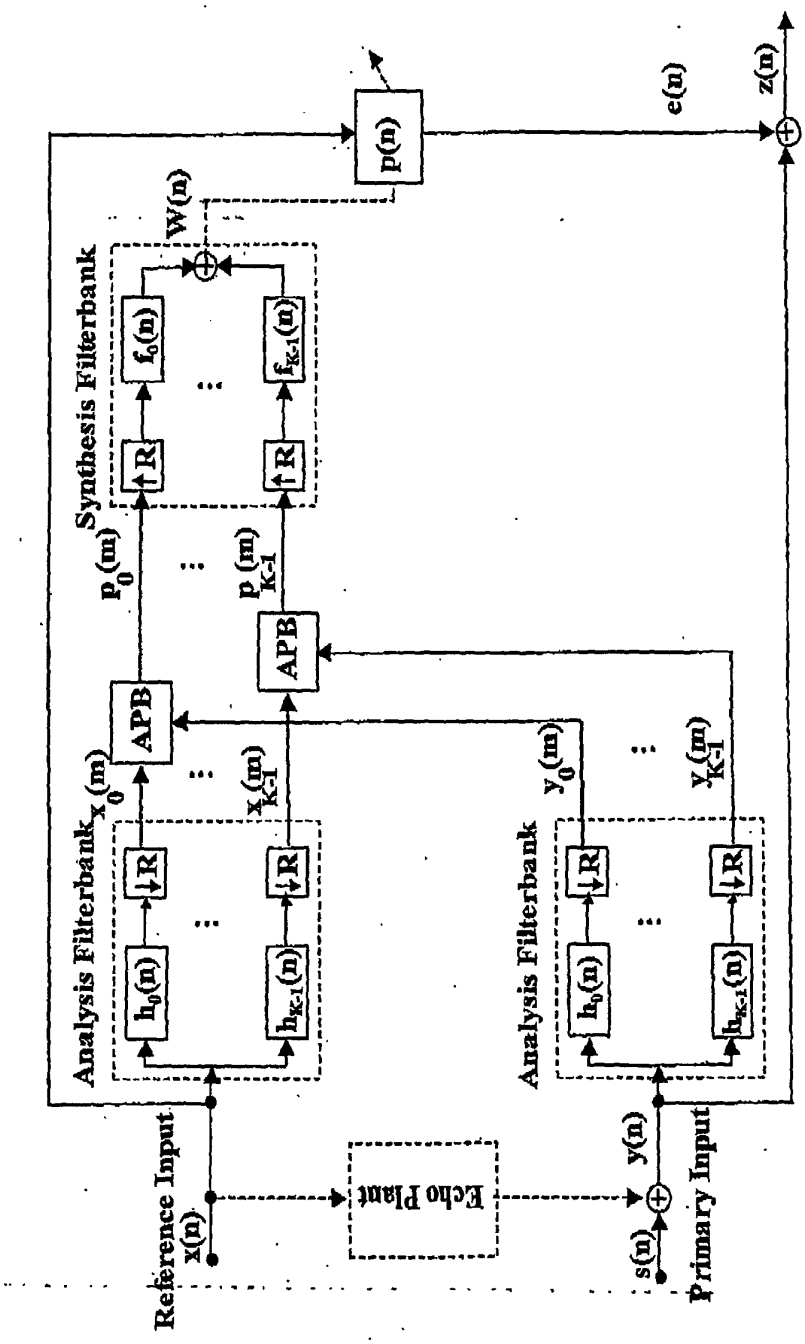


Fig. 24 Reconstruction of TAF through WOLA synthesis of the SAFs

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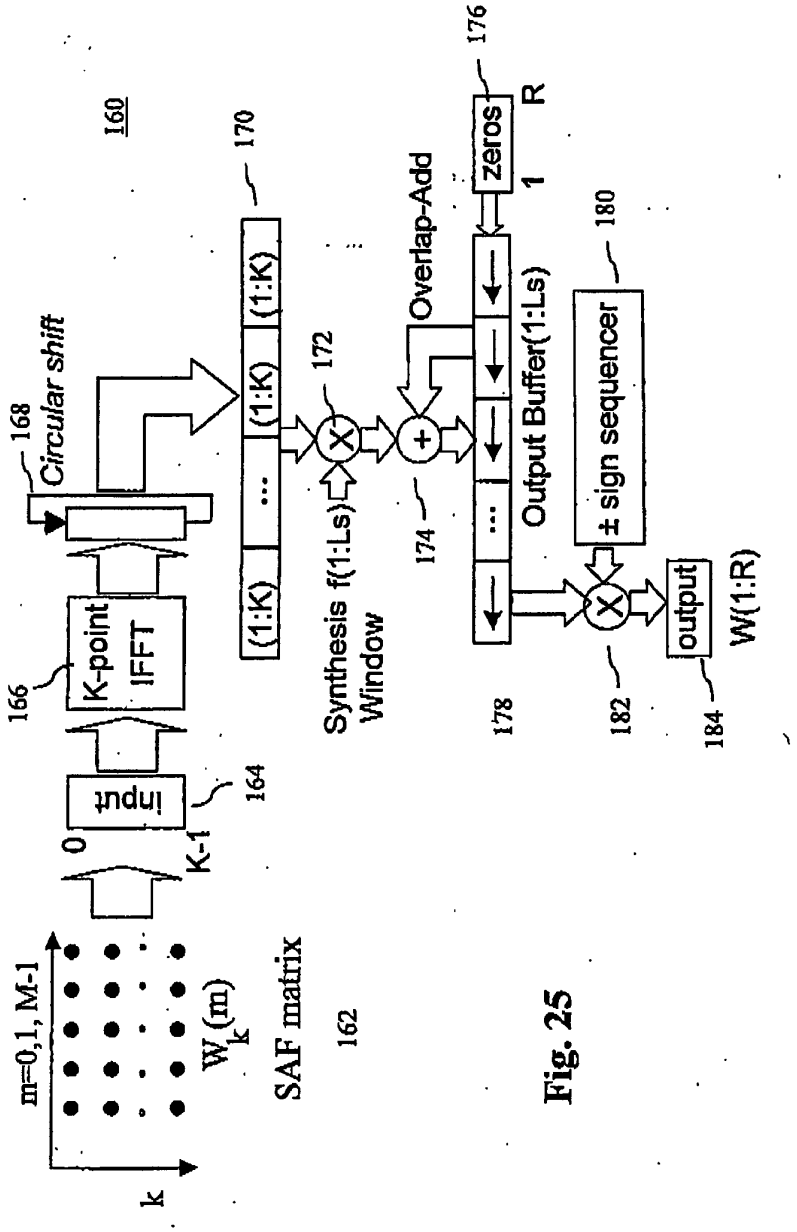

160
168
Circular shift
166
K-point IFFT
input
164
0
K-1
m=0,1, M-1
k
$W_k(m)$
SAF matrix
162
170
(1:K)
(1:K)
...
(1:K)
172
Synthesis f(1:Ls) Window
174
Overlap-Add
176
zeros
1
R
Output Buffer(1:Ls)
178
180
± sign sequencer
182
output
184
W(1:R)

Fig. 25

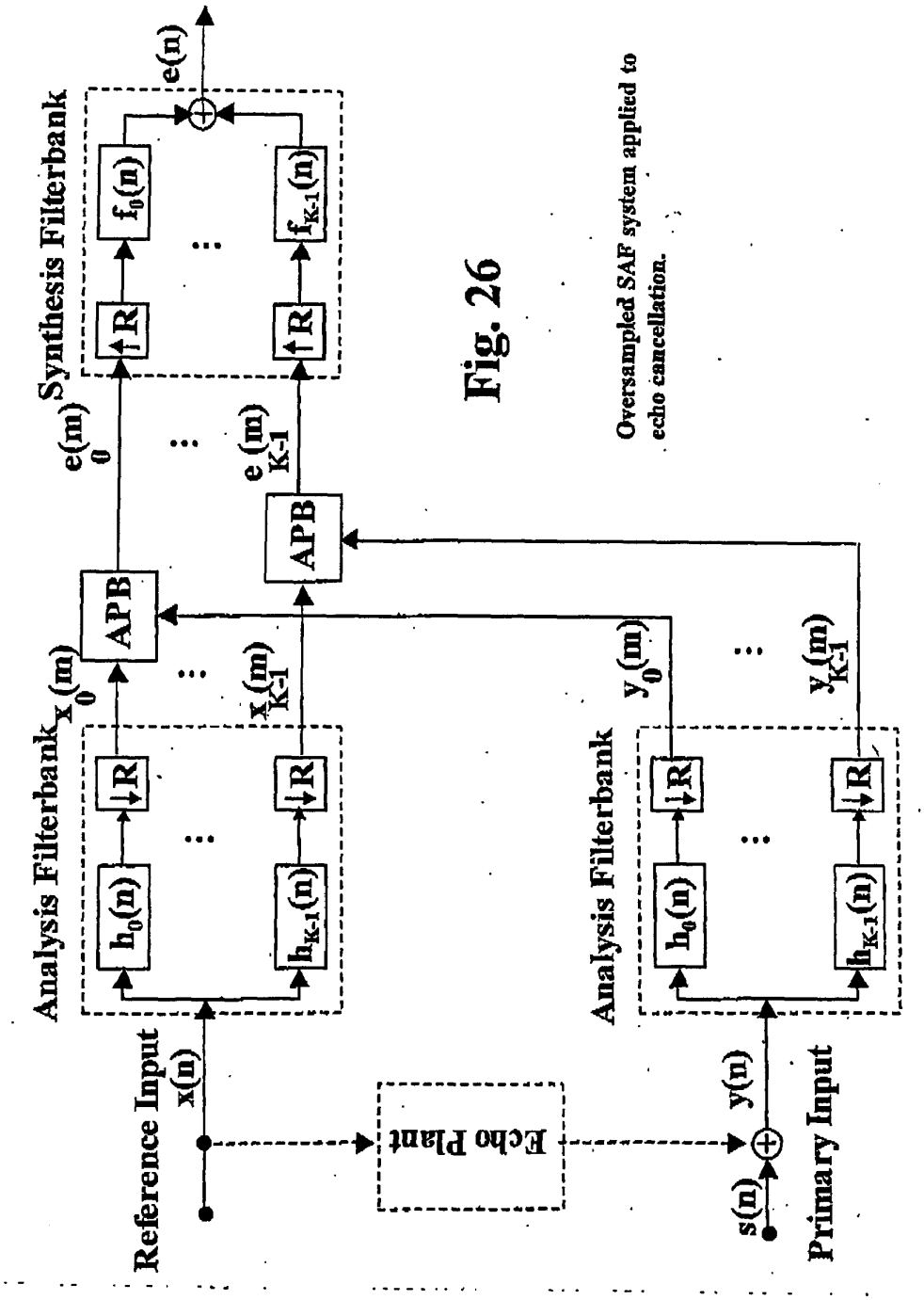


Fig. 26

Oversampled SAF system applied to echo cancellation.

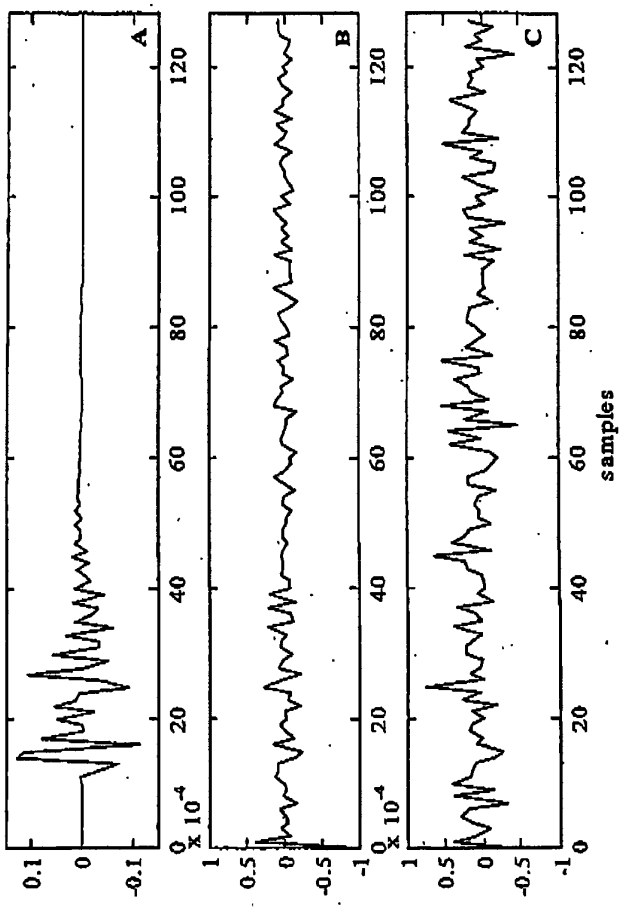


FIG. 27

(A) Time-domain echo plant, and the reconstructed plant with SGS-PAP adaptation, for D=1,8 ,

(B) error in reconstructed TAFs for D=1 and (C) error for D=8.

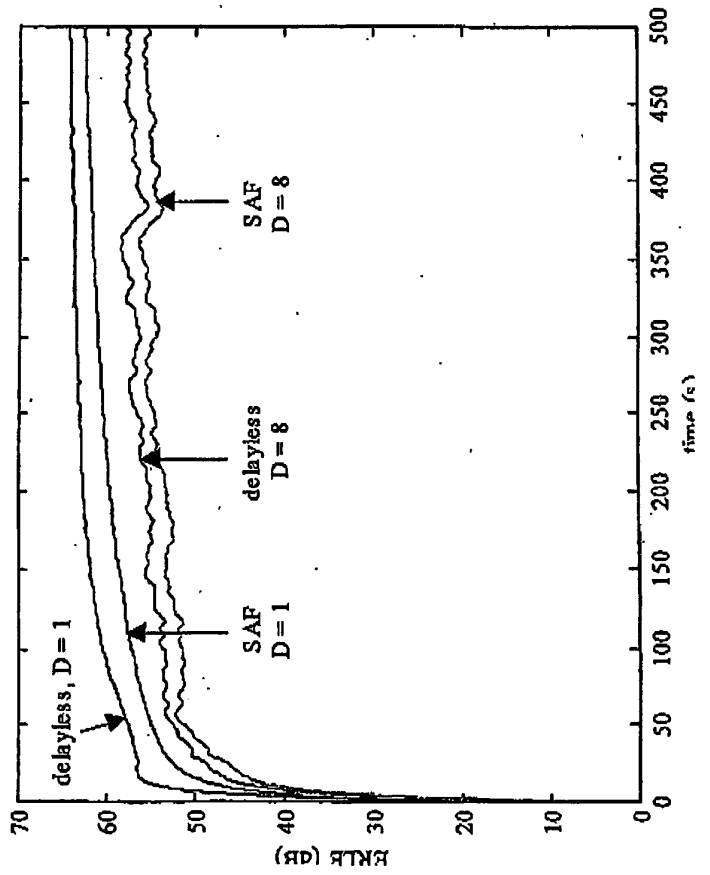


FIG. 28 ERLE results for SGS-PAP adaptation with D=1,8, for Figure 24 and 26.

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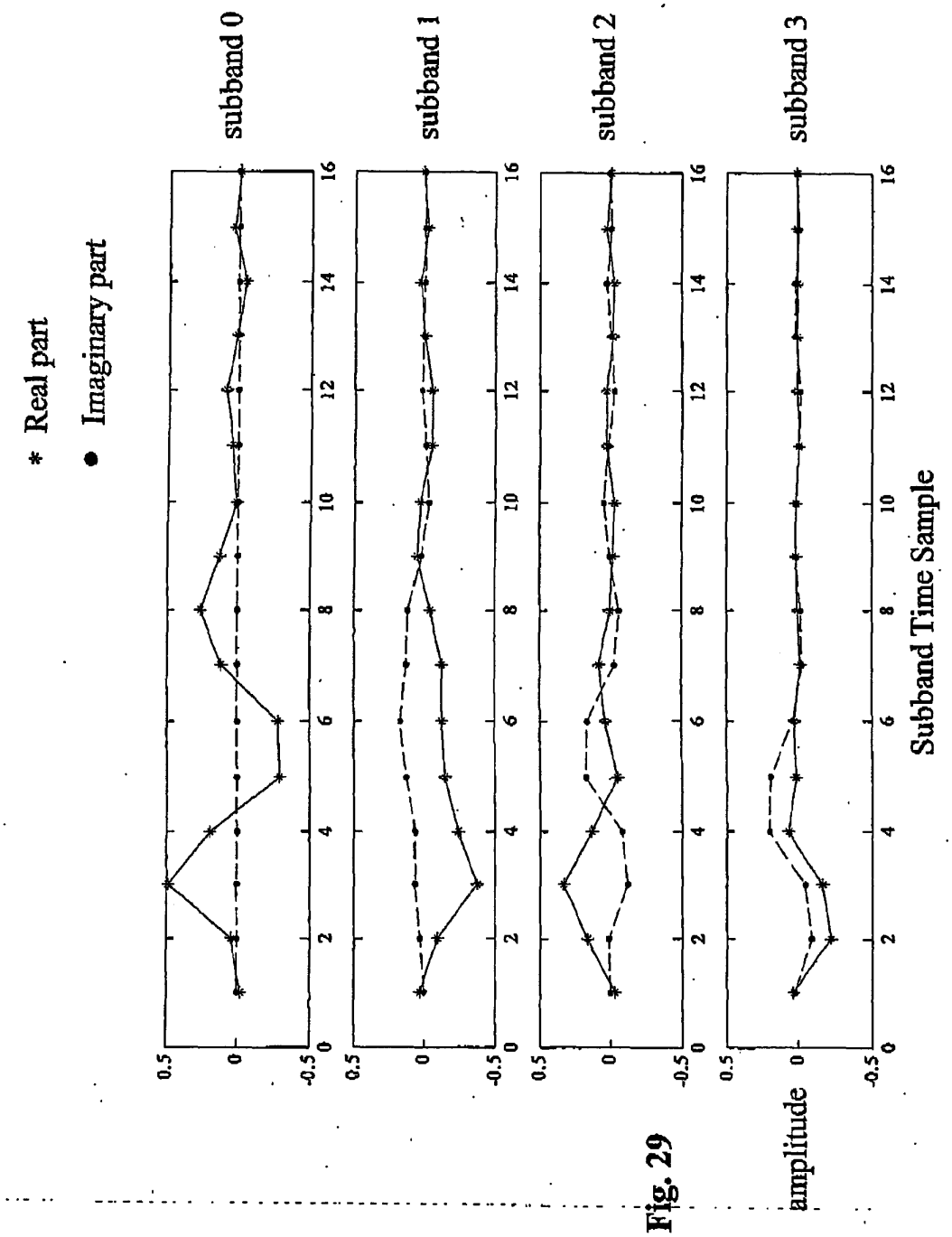

* Real part
• Imaginary part
subband 0
subband 1
subband 2
subband 3
amplitude
Subband Time Sample

Fig. 29

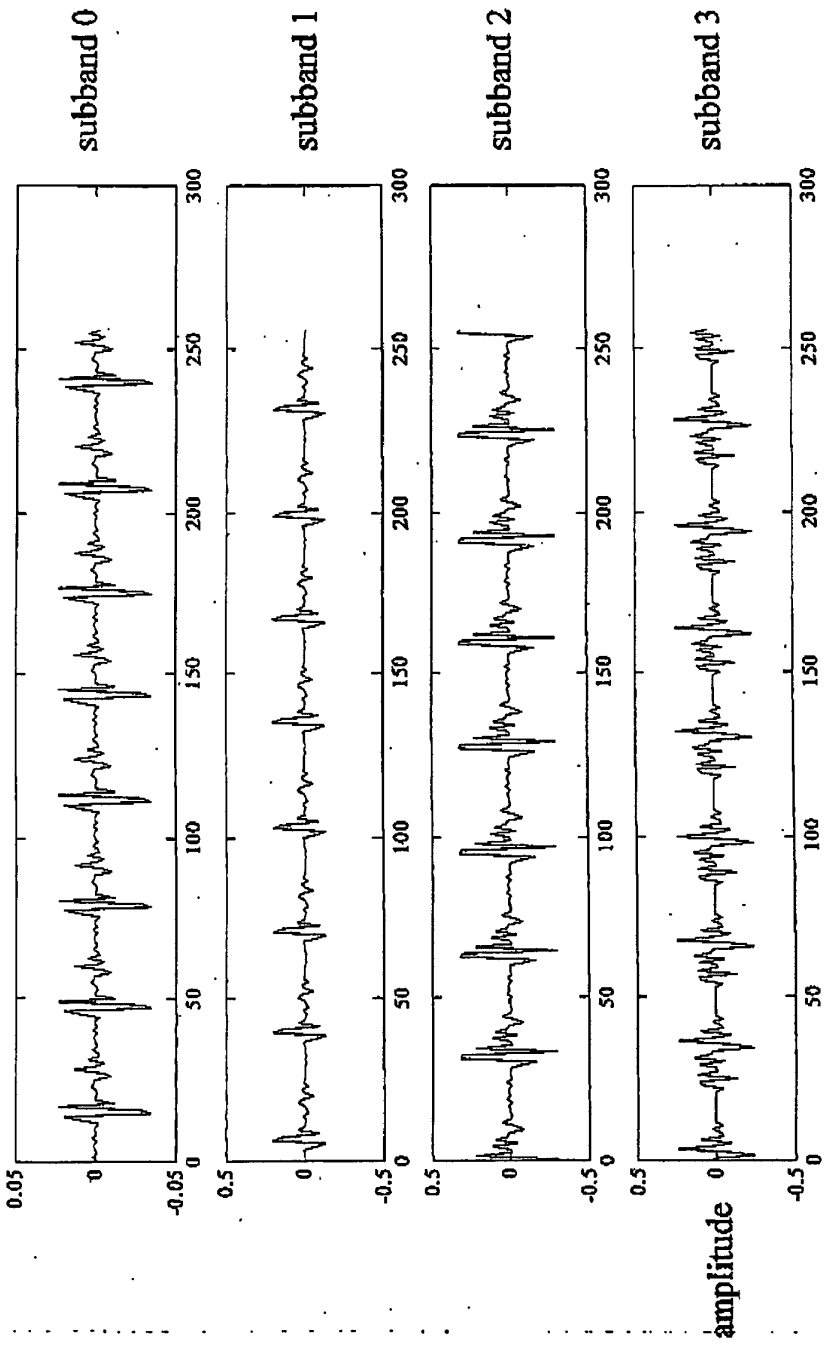

subband 0
subband 1
subband 2
subband 3
amplitude
Time Sample

Fig. 30

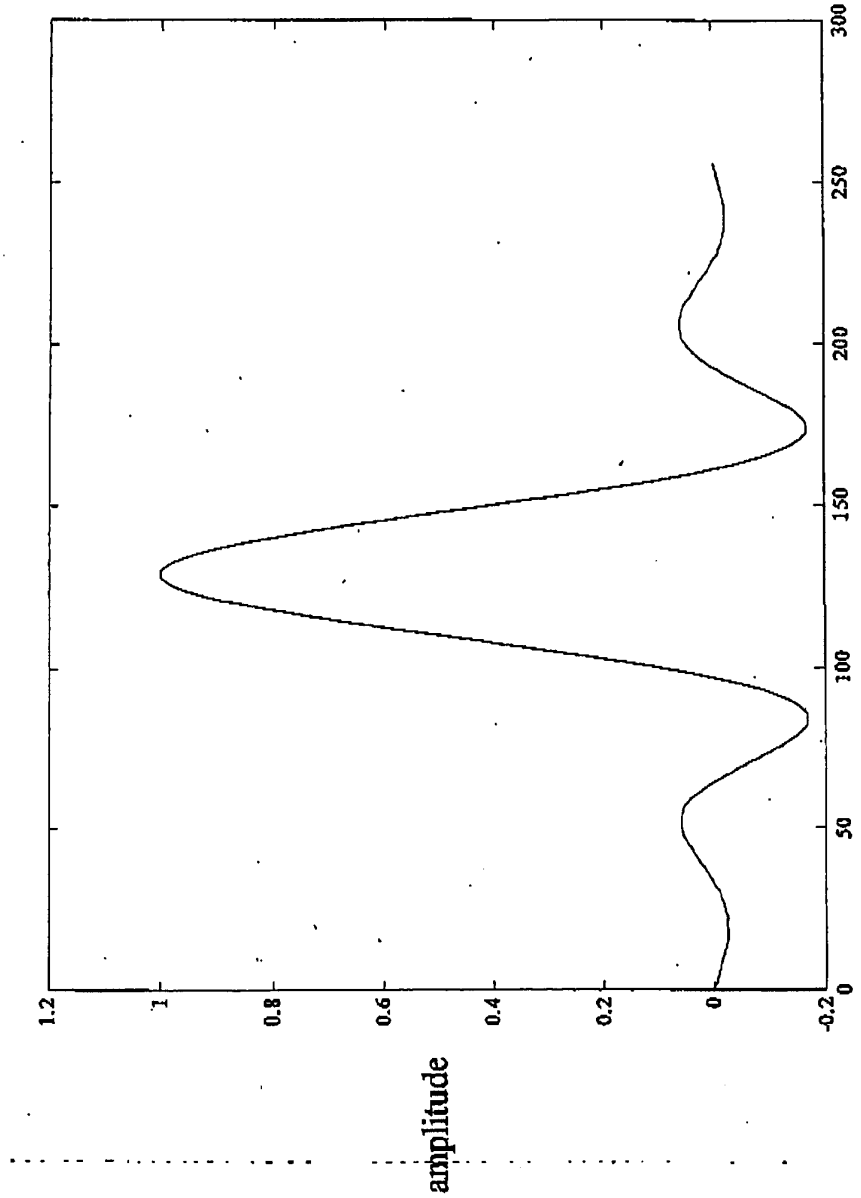

amplitude
1.2
1
0.8
0.6
0.4
0.2
0
-0.2
0
50
100
150
200
250
300
Time Sample

Fig. 31

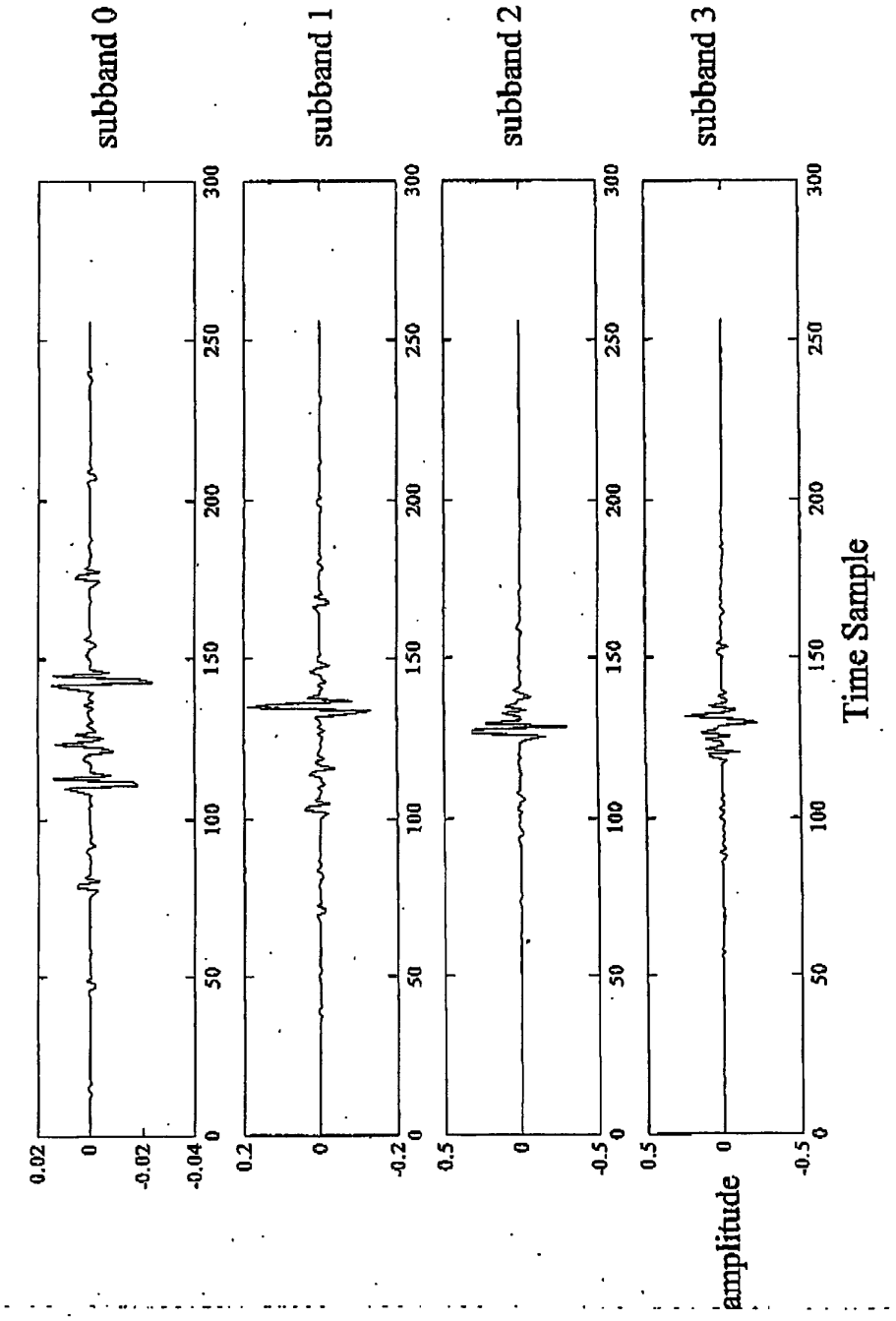

subband 0
subband 1
subband 2
subband 3
amplitude
Time Sample

Fig.32

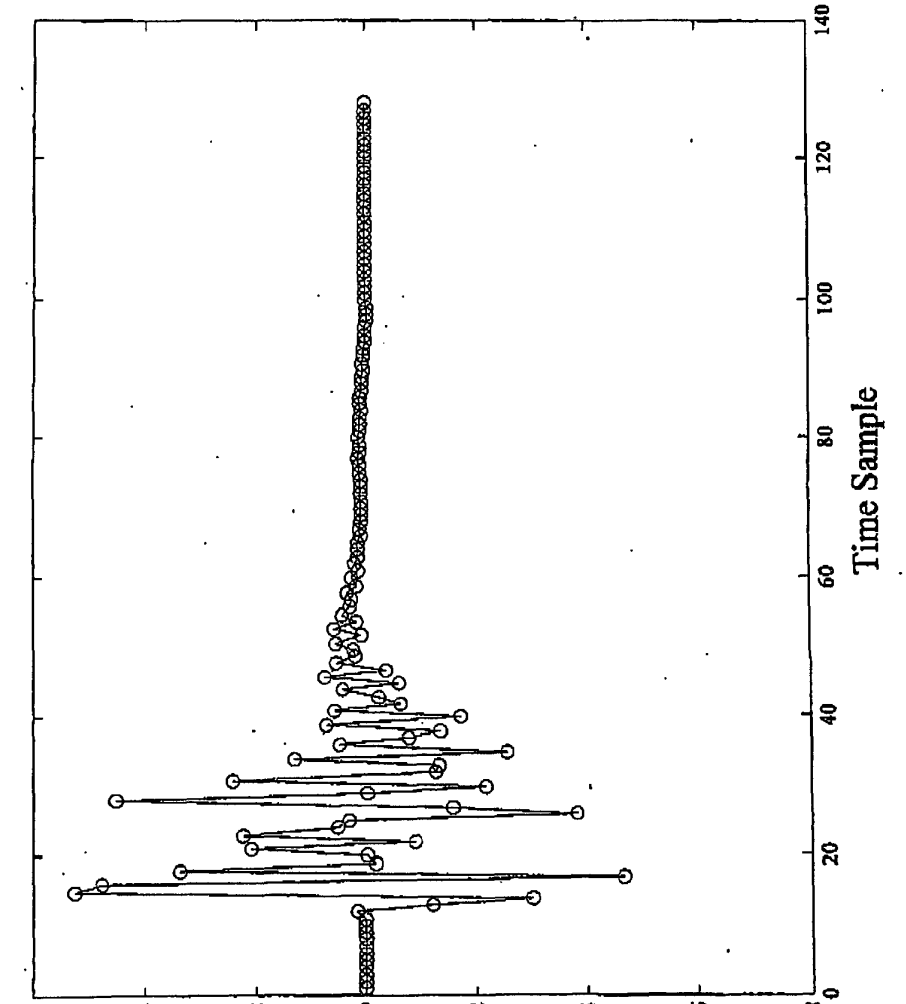

amplitude
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
Time Sample
0
20
40
60
80
100
120
140

Fig. 33

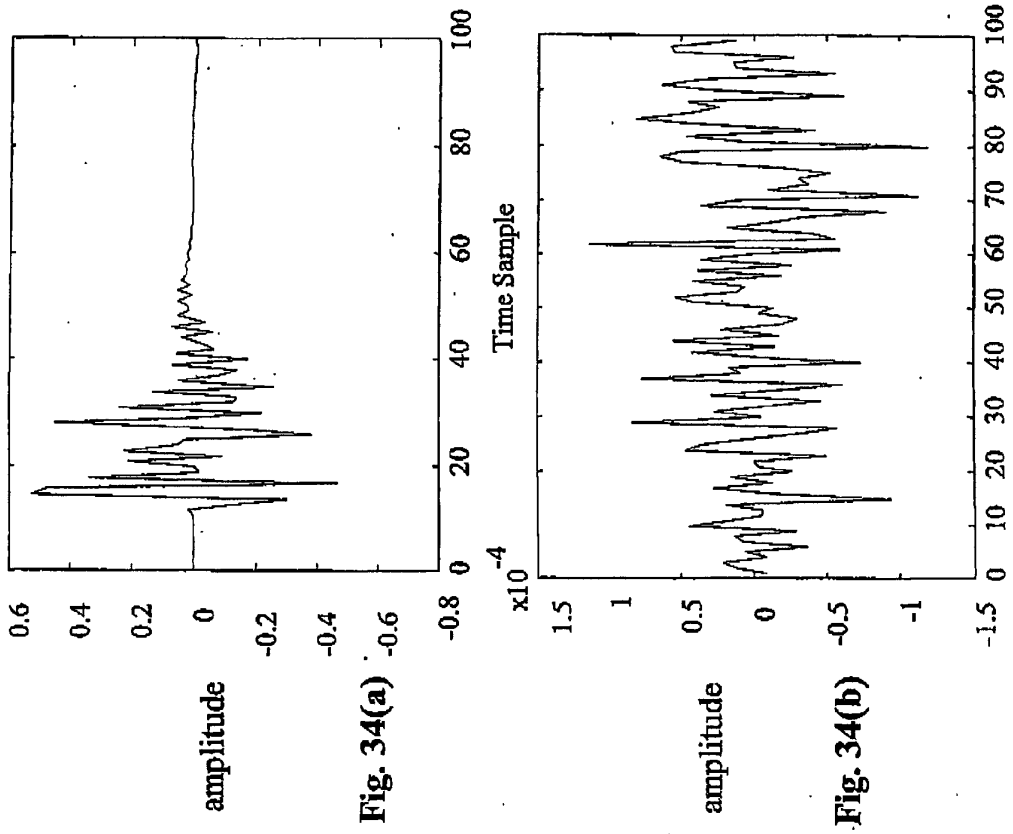


Fig. 34(a)

Fig. 34(b)

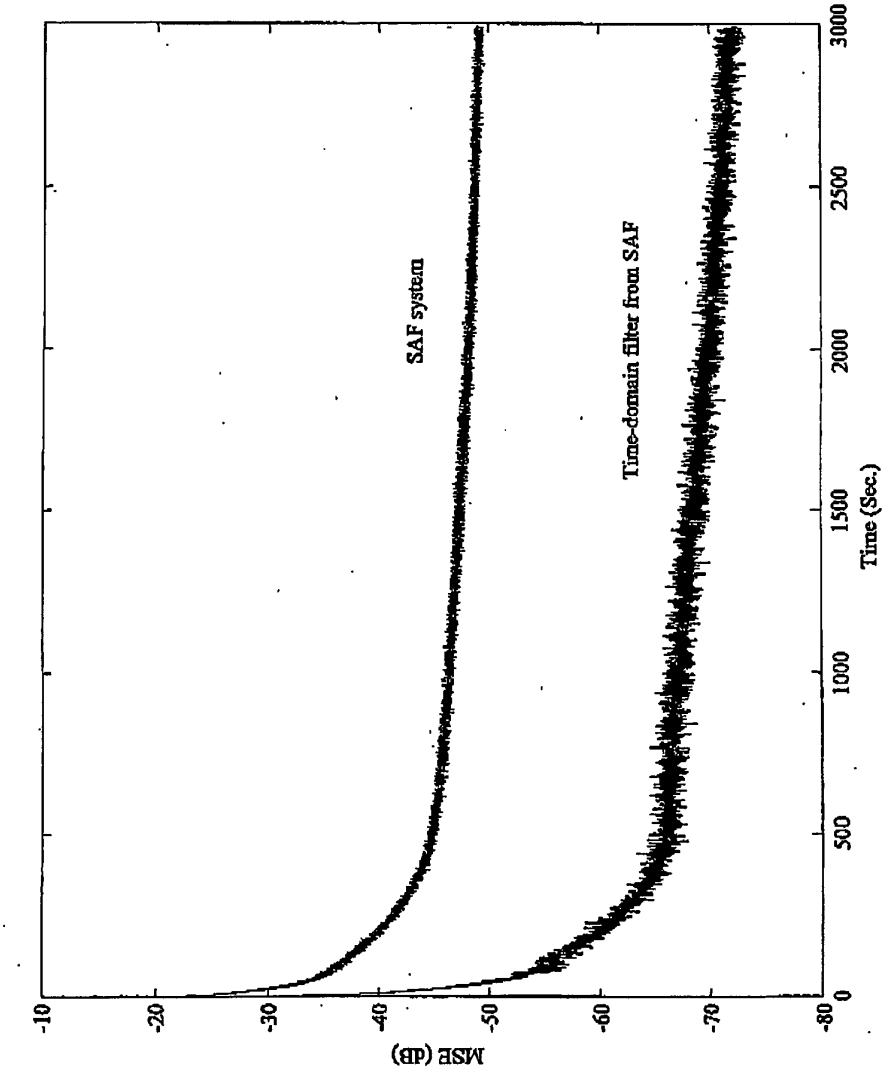

SAF system
Time-domain filter from SAF
MSE (dB)
Time (Sec.)
-10
-20
-30
-40
-50
-60
-70
-80
0
500
1000
1500
2000
2500
3000

Fig. 35

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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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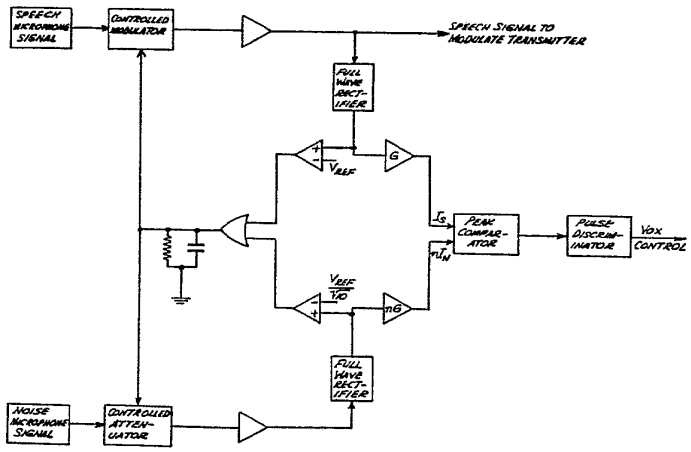
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(81) Designated States: AT (European patent), AU, BE (European patent), CH (European patent), DE (European patent), DK, FR (European patent), GB (European patent), IT (European patent), JP, LU (European patent), NL (European patent), SE (European patent), US.

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(54) Title: ADAPTIVE SPEECH DETECTOR SYSTEM



(57) Abstract

A speech detector circuit for selectively transmitting speech signals while suppressing ambient sound related signals comprises an input from a microphone producing a speech signal and an input from a microphone producing a noise signal. The circuit includes a comparator which compares the speech signal with peak values of the noise signal to produce a pulse signal which indicates when the speech signal exceeds a threshold and a discriminator responsive to the pulse signal to produce a detected speech signal when the pulse signal occurs within a selected period which is spaced from a preceding pulse signal by a selected period.

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ADAPTIVE SPEECH DETECTOR SYSTEM

TECHNICAL FIELD OF THE INVENTION

This invention relates to speech detector systems. The invention is particularly suitable for use in combination with a transmitter for voice communication and which is equipped with a dual-microphone amplifier system arranged so as selectively to transmit speech signals whilst suppressing ambient-sound-related signals.

There are a number of applications for speech detector systems. For example, wireless transmitters, especially of the portable type, are commonly required to have a VOX facility, that is, automatic keying of the transmitter in response to speech signals.

BACKGROUND ART

In previously-known VOX systems which utilise a single microphone channel, it has been difficult to combine effective recognition of the operator's speech with adequate rejection of unwanted background noise or speech. Various techniques have been employed to improve the voice discrimination, including band-separation filtering and the use of

noise-cancelling microphones coupled with automatic gain control, abbreviated as "AGC". However, where a wide amplitude-dynamic-range is achieved by the use of AGC it is difficult to avoid false-triggering of the VOX circuit. In addition, the AGC system tends to amplify background noise signals and circuit noise to an objectionable level during periods when the speech is interrupted. This is especially evident when a transmitter is switched to a manual-keying or press-to-talk mode, sometimes abbreviated as "PTT".

A system which has been used successfully to overcome the foregoing disadvantages of VOX and AGC systems comprises a two-channel dual-microphone arrangement in which one microphone receives the operator's speech, superimposed on ambient noise, and the other principally receives ambient noise. Not only is noise cancellation possible, over at least the lower part of the frequency range, but the ambient noise signal may be used to control the speech channel gain in the absence of speech and, by this method, prevent the noise signal at the speech channel output from rising above, for example, a level 10dB below the nominal speech signal output level. A significant advantage of this system when combined with a speech detector is that, at least in the steady state, there is a well-defined difference in speech and noise levels which is easily discriminated in a simple comparator circuit.

Some constraints and disadvantages of the dual-microphone system are:

A time delay must be included in the speech detector response to take account of the AGC attack time which can be, for example, up to 5 ms.

The two microphones must be closely matched in sensitivity and frequency response.

Direct noise cancellation is usually only possible at frequencies up to about 500 Hz because of the transit time difference for a sound pressure wave travelling to each of the two microphones, this varying with the direction of the sound source.

There is a variation of the automatic-gain-controlled ouput level of either channel with input signal level, depending on the loop gain of the AGC loop.

There is an uncertainty in the output reference level of the AGC system because of production tolerances in components.

There is a need for precise tracking, or equality of gain, in the control elements of the speech and noise channels.

There is an uncertainty in the speech detector reference level owing to production tolerances in components.

A number of the constraints and disadvantages listed above can produce effects which are additive.

This can give rise to a substantial error in the effective speech detector threshold. Even with the inclusion in the circuit of a speech detector threshold adjustment means to take account of static mismatch with the preceding microphone and AGC amplifier system, dynamic uncertainties may still give rise to false-triggering by the speech detector.

An object of the present invention is to provide circuit means which at least ameliorate some of the above disadvantages of the prior art and which in preferred embodiments reduce if not eliminate the effects of variations in AGC levels, mismatch between speech detector and AGC reference levels, tracking errors of AGC elements and transient signals during the AGC attack time. In principle this is achieved by combining the functions of AGC and speech detection to eliminate mismatch occurring in separate circuits and, further, by dynamically comparing the detected speech level with the detected noise level instead of with a constant threshold.

DISCLOSURE OF INVENTION

This invention consists in a speech detector system comprising means for producing a first (speech) signal representative of speech superimposed on ambient noise and a second (noise) signal representative of said ambient noise;

comparator means to determine a speech signal

threshold according to peak values of said second signal and to produce a third (pulse) signal which provides an indication of when said first signal exceeds said speech signal threshold; and

discriminator means responsive to an indication by said third signal to produce a fourth signal indicating detected speech when an indication occurs within a second selected period which is spaced from a preceeding indication by a first selected period.

For preference the speech signal and noise signal are subjected to automatic gain control and noise cancellation prior to input to the comparison means. For preference also the ratio of the noise cancelled speech signal to the noise signal fed to the comparison means is greater than 1:1.

Other aspects of the invention will be apparent from the description which follows.

BRIEF DESCRIPTION OF DRAWINGS

An embodiment of the invention will now be described by way of example only with reference to the accompanying drawings wherein:

Figure 1 is a schematic block diagram showing part of a wireless transmitter speech detector system having a two channel audio system with speech and noise inputs.

Figure 2 shows a first circuit suitable for use as the peak comparator shown as a block in Fig. 1.

Figure 3 shows a circuit suitable for use as the

pulse discriminator shown as a block in Fig. 1.

Figure 4 shows a preferred circuit which combines the peak comparator and pulse discriminator in a single circuit.

Figure 5 shows a further preferred circuit which combine the peak comparator and pulse discriminator.

BEST MODE FOR CARRYING OUT INVENTION

The block diagram of Figure 1 shows schematically a wireless transmitter speech detector system having a two channel audio system with speech and noise inputs, and AGC. The rectified speech and noise signals are compared to detect the presence of speech.

The AGC system operates in substantially the known manner to control the peak noise level at the output of the noise channel, to a predetermined value equal to $V_{REF}/\sqrt{10}$. The high-frequency components of noise in the speech channel, which are not removed by the low-frequency noise-cancellation circuit, will generally be of similar amplitude to the output of the noise channel. The foregoing control mode is over-ridden when the level of signal in the speech channel output exceeds V_{REF}, thus suppressing the noise further and enabling an effective signal-to-noise ratio for speech of 10dB to be maintained in noisy environments.

It has been found that, in a practical system, it is desirable to have the VOX control switched at a speech-to-noise ratio of approximately 6dB, based on

peak values. In previously-known speech detectors it has been usual, therefore, to detect, by means of a comparator amplifier, the instances when a signal in the speech channel exceeds a predetermined value of $\approx 2V_{REF}/\sqrt{10}$.

In the present invention, the peak value of the rectified noise-channel signal is itself used, after amplification, as the comparator reference as shown in Fig. 2.

Figure 2 shows a peak comparator circuit for use in the arrangement shown in Figure 1. Rectified speech signal I_s representative of speech superimposed on ambient noise (a first signal) and rectified noise signal nI_N representative of ambient noise (a second signal) are fed to the inputs as shown after amplification by amplifiers shown in Figure 1, where n is the ratio of the amplification of the noise channel to the amplification of the speech channel.

The current signals I_s and nI_N respectively generate voltages V_s and V_N across the input resistors R_1 and R_2.

Voltage V_N generated by signal nI_N appears at the base of transistor Q1 which has its collector connected to a positive power supply V_{CC}. The emitter of transistor Q1 is connected to an R-C combination of resistor R4 and capacitor C2. Transistor Q1, capacitor C2 and resistor R4 comprise peak detecting means. The

transistor Q1 operates as a voltage follower when V_N is greater than the voltage on capacitor C2 and is switched off when V_N is less than the voltage on capacitor C2. Current to charge capacitor C2 is drawn from supply V_{CC}. The charging current is therefore independent of the current drawn by resistor R2 thereby avoiding response lag. Resistor R4 is in parallel with capacitor C2 and the decay time constant R4C2 is long enough for acceptable smoothing of random noise.

Voltage V_s generated by signal I_s charges capacitor C1 through diode D1 only when the voltage stored across capacitor C1 is less than V_s. That is, when a positive pulse or peak occurs in the speech signal I_s. The time constant of resistor R1 and Capacitor C1 is short enough to allow C1 to be charged by peaks in I_s corresponding to speech "glottal" pulses, but long enough to filter out transient noise. Resistor R3 is in parallel to capacitor C1 and its value sets the decay time of the charge on capacitor C1. The decay time constant resistor R3 and capacitor C1 is made equal to the time constant of resistor R4 and capacitor C2 to provide good dynamic tracking.

The voltages appearing across capacitors C1 and C2 are fed to the inputs of a differential amplifier OP1 which produces a positive output when the voltage across C1 exceeds the voltage across C2 and a negative output in the reverse situation. That is, the voltage across

capacitor C2 constitutes a speech signal threshold and the output of amplifier OP1 comprises a third signal which provides an indication of when the speech signal exceeds the speech signal threshold.

The rectified noise signal amplification in Figure 1 is usually twice the rectified speech signal amplification so that a positive peak in the speech signal of at least twice the level of ambient noise is required to produce a positive output from differential amplifier OP1.

Fig. 3 shows one kind of pulse discriminator circuit, the principle purpose of which is to generate an output VOX control pulse only if a series of speech pulses is received but not when a single pulse or short burst of pulses is received. This is achieved by discriminating between glottal pulses comprising speech, which typically occur at 6-8 ms intervals, and single pulses or short bursts of pulses separated by less than 3 ms generated by noise. In this way the system provides immunity to transient noise such as that caused by an impact, which is typically too fast for AGC response to be effective.

In the system shown, a bistable latch is used to generate a VOX control or "speech detected" signal. When a trigger pulse is received from the peak comparator, a timer comprising a double pulse generator generates two control pulses, A and B, with a first

selected period T1 and a second selected period T2 respectively. Control pulse A and the trigger pulse from the peak comparator are fed to an AND gate. The output of the AND gate is connected to the SET input of the bistable latch. A delay is included in the trigger pulse connection to prevent a trigger pulse reaching the AND gate before control pulse A. In this way control pulse A inhibits the setting of the bistable latch by either the initial trigger pulse or any subsequent pulse occurring within the period T1, typically 5 ms. However, a trigger pulse occurring within the period T2 is able to set the bistable latch as well as re-starting the timing period T2 of control pulse B. The latter function ensures that the output VOX control pulse has at least a period equal to the initial or minimum value of T2, 10 ms for example. This period is determined by the requirements of any ensuing transmitter circuit. At the end of the period T2 after the last trigger pulse, the bistable latch is reset and control pulse A trigger is enabled.

Fig. 4 shows a system which combines the peak comparator and pulse-discriminator functions in one circuit, resulting in a substantial saving in components which can be important in the envisaged applications.

The transistors Q1 and Q2 act as peak detecting means for the noise and speech rectified inputs respectively, and because transistors Q1 and Q2 share a

common connection to ground via the parallel combination of resistor R4 and a storage element comprising capacitor C2, the collector-emitter current of either transistor on input signal peaks is dependent on the previous peak value to which capacitor C2 has been charged. By suitable choice of the decay time of capacitor C2 via resistor R4, the transistor pair Q1, Q2 can therefore also act to supress second and subsequent peaks separated by less than the normal "glottal" period.

This is achieved because the total charge flowing into the collector of transistor Q2 during a detected speech pulse input I_S is dependent on the instantaneous charge of capacitor C2, which is arranged to decay via resistor R4. It follows that, if a rapid burst of pulses of similar amplitude occur in I_s, capacitor C2 will charge up rapidly and only one or two large pulses of charge will flow into the collector of transistor Q2, followed by small charge pulses which are sufficient to keep capacitor C2 charged. In contrast, speech signals typically have large impulses, or glottal pulses, spaced apart at 6-8 ms intervals with smaller amplitude pulses in between. By choice of decay time capacitor C2 charge decays enough for each glottal pulse to produce a large current impulse in the collector lead of transistor Q2. It is therefore quite a simple matter to discriminate between speech and either continuous or impact noise by integrating the collector charge

impulses and comparing this voltage with a suitable reference value using a Schmitt trigger circuit.

Integration is achieved in the circuit of Figure 4 by means of the R-C combination comprising resistor R5 and capacitor C3. Charge impulses from the collector lead of transistor Q2 result in a voltage appearing across capacitor C3 with respect to constant voltage supply V_{cc}. The reference voltage is chosen so that it is exceeded by the voltage across C3 when three consecutive pulses occur in the collector lead of transistor Q2 without substantial decay of the charge on capacitor C3 between pulses. This is achieved by suitable choice of resistor R5 so that "staircase" integration of current pulses is obtained for pulses occurring within a selected period.

The Schmitt trigger is preferred to a comparator, for example, to ensure clean switching and an output pulse width in excess of some arbitrary value, dependent on the needs of ensuing circuitry. This is related partly to the decay time-constant of capacitor C3 which is controlled by resistor R5. The decay time-constant is principally determined by the need for recovery between typical impact-noise occurrences, but must be long enough for effective "staircase" integration of glottal pulses. A resistor R6 is included in the collector lead of transistor Q2 to limit the amplitude of charge impulses in the event of the AGC system being

overloaded momentarily by a large impact-noise signal, thus ensuring that effective discrimination is maintained.

As will be apparent to those skilled in the art, the amplification of detected noise inputs I_N by a greater factor n than speech input I_s further enhances the discrimination of the circuit by, firstly, generating a comparator threshold proportional to the ambient noise level and thereby reducing the frequency of Q2 collector impulses resulting from random noise and, secondly, because of the previously-discussed properties of the two-microphone system, permitting operator speech to be distinguished from more distant "ambient" speech, the effectiveness of the approach being improved because of the reduced reliance on AGC control accuracy.

Figure 5 shows a further circuit combining the peak comparator and pulse discriminator functions. The circuit operates in substantially the same manner as the circuit of Figure 4 however the transistors Q4 and Q5 act as a current mirror in a known manner to produce a current flowing through collector of transistor Q5 proportional to the collector current of transistor Q2. This allows R-C combination of resistor R5 and capacitor C3 to be connected to ground thereby eliminating the need for maintaining the voltage supply V_{cc} in Figure 4 constant.

It will be apparent that resistor R5 and capacitor C3 effectively integrate charge impulses flowing from transistor Q5 collector in substantially the same manner as described above to control the voltage at the input of Schmitt trigger S1 which generates a VOX control or "speech detected" signal when a predetermined threshold voltage is reached.

The amplitude of charge impulses from the collector of Q5 is limited by the shunting action of Q3 in response to the voltage developed across R7 by the current flowing into Q2 collector.

As will be apparent to those skilled in the art the invention hereof may be embodied in other circuits which function in an equivalent or analagous manner and such embodiments are within the scope hereof.

THE CLAIMS DEFINING THE INVENTION ARE AS FOLLOWS:-

1. A speech detector system comprising means for producing a first (speech) signal representative of speech superimposed on ambient noise and a second (noise) signal representative of said ambient noise;

comparator means to determine a speech signal threshold according to peak values of said second signal and to produce a third (pulse) signal which provides an indication of when said first signal exceeds said speech signal threshold; and

discriminator means responsive to an indication by said third signal to produce a fourth signal indicating detected speech when an indication occurs within a second selected period which is spaced from a preceding indication by a first selected period.

2. A speech detector system as claimed in claim 1 wherein said speech signal threshold is determined from said second signal by peak detecting means including a storage element.

3. A speech detector system as claimed in claim 2 wherein said first signal is compared with said speech signal threshold by means of a differential amplifier, the output of the differential amplifier producing said third signal.

4. A speech detector system as claimed in claim 2 wherein said comparator means comprises a three terminal amplifier to generate said third signal, said amplifier

having the terminal common to input and output connected to the storage element of said peak detector means.

5. A speech detector system as claimed in claim 4 wherein current flow through the output terminal of said three terminal amplifier comprises said third signal.

6. A speech detector system as claimed in claim 5 wherein said 3-terminal amplifier is a transistor and said terminal common to input and output is the emitter.

7. A speech detector system as claimed in claim 4 or claim 5 further comprising means to limit said third signal amplitude.

8. A speech detector system as claimed in claim 5 or 6 further comprising a current mirror to produce a mirrored current flow proportional to said third signal current flow and wherein said discriminator means is responsive to the mirrored current flow.

9. A speech detector as claimed in claim 8 further comprising transistor current shunting means to limit said mirrored current flow.

10. A speech detector system as claimed in any one of claims 1 to 9 wherein said discriminator means comprises a control pulse generator which in response to an indication in said third signal generates a first control pulse having a duration equal to said first selected period and a second control pulse having a duration equal to said second selected period, said first control pulse acting to inhibit the production of

a fourth signal during the duration thereof and said second control pulse acting to permit production of a fourth signal only in response to pulse indications in said third signal occuring during the duration of said second control pulse.

11. A speech detector system as claimed in claim 10 wherein a bistable latch generates said fourth signal and said first and second control pulses act to control operation of the bistable latch by said third signal.

12. A speech detector system as claimed in any one of claims 1 to 9 wherein said discriminator means comprises an integrator the output of which controls production of said fourth signal, said second selected period being determined by the integration constant.

13. A speech detector system as claimed in claim 12 wherein the output of the integrator is compared with a reference voltage to produce said fourth signal.

14. A speech detector system as claimed in any one of claims 11 to 13 wherein the integrator comprises a capacitor and the integration constant is determined by the decay time of the capacitor through a parallel connected resistance.

15. A speech detector system as claimed in any of claims 1 to 14 wherein said first signal and said second signal are amplified and the second signal is amplified by a greater amount than the first signal.

16. A speech detector system substantially as herein

described with reference to Figures 1, 2 and 3, or Figures 1 and 4, or Figures 1 and 5 of the accompanying drawings.

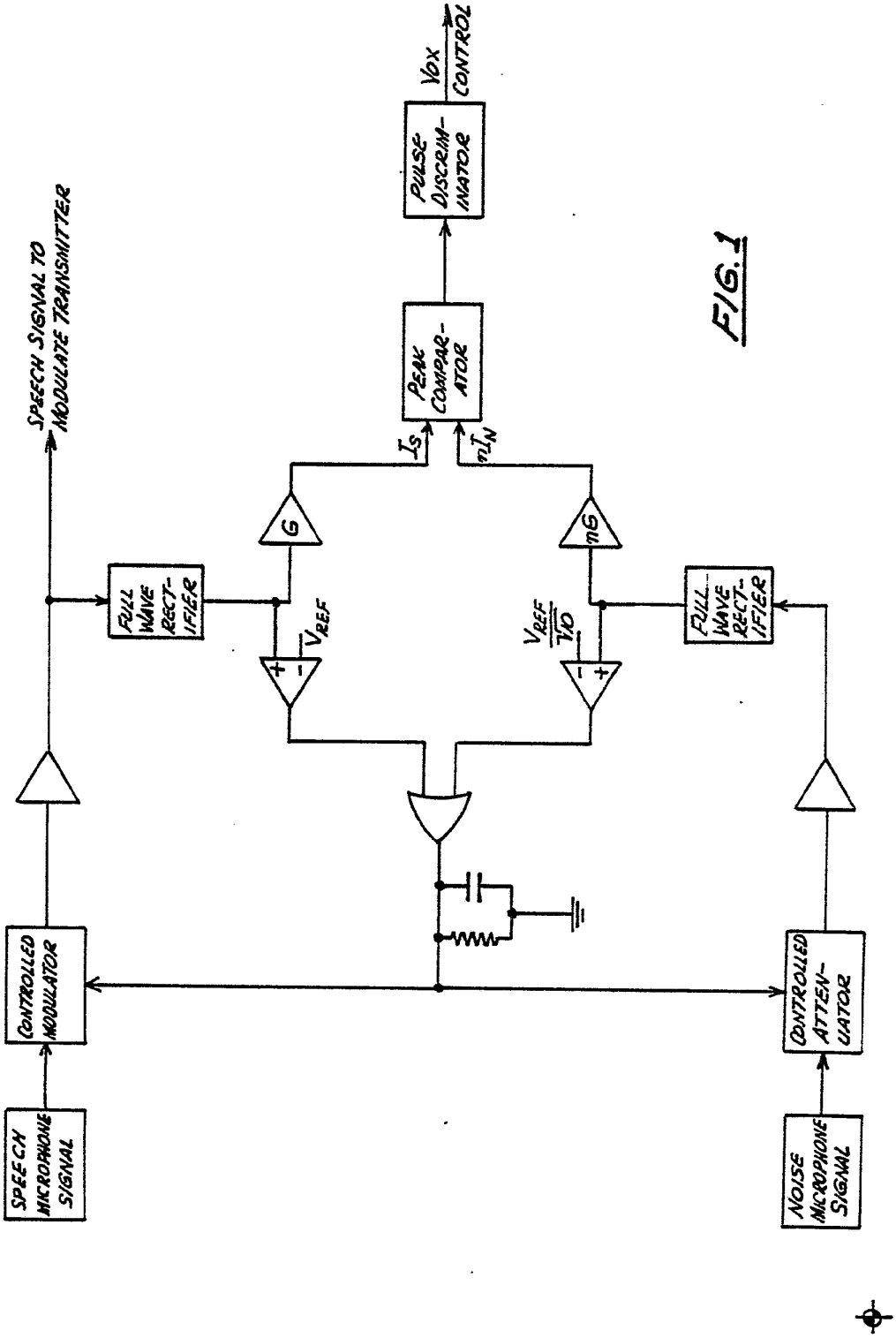

SPEECH SIGNAL TO MODULATE TRANSMITTER
PULSE DISCRIM-INATOR
VOX CONTROL
PEAK COMPAR-ATOR
I_S
nI_N
G
nG
FULL WAVE RECT-IFIER
FULL WAVE RECT-IFIER
V_{REF}
$\frac{V_{REF}}{1/0}$
CONTROLLED MODULATOR
CONTROLLED ATTEN-UATOR
SPEECH MICROPHONE SIGNAL
NOISE MICROPHONE SIGNAL

FIG. 1

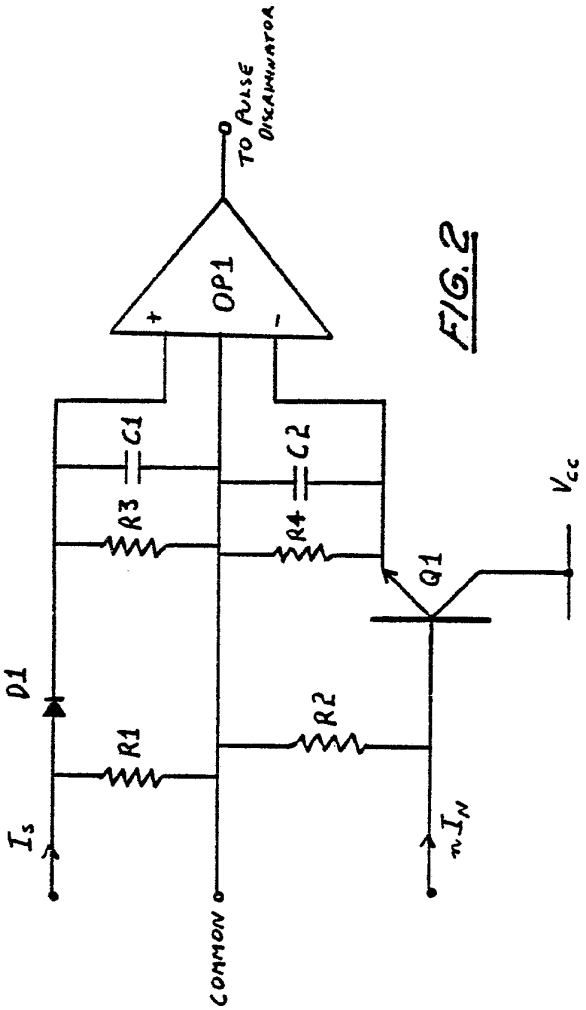

TO PULSE DISCRIMINATOR
OP1
+
-
C1
C2
R3
R4
Q1
V_{cc}
D1
R1
R2
I_s
nI_N
COMMON

FIG. 2

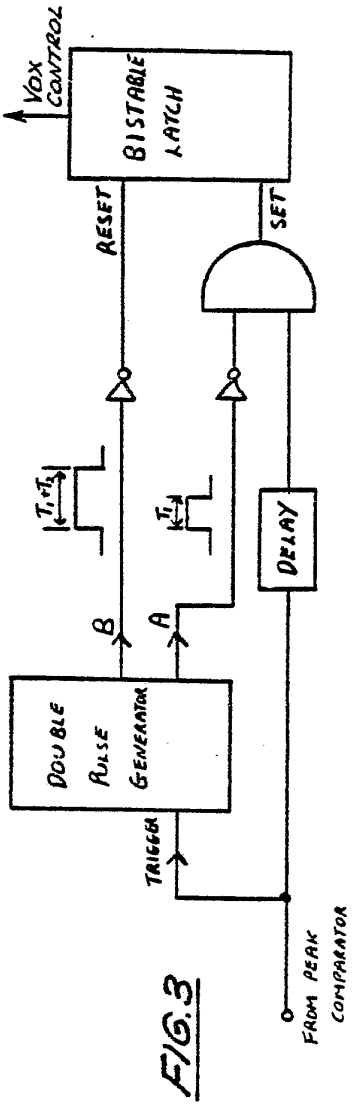

VOX CONTROL
BISTABLE LATCH
RESET
SET
T_1+T_2
T_1
DELAY
B
A
DOUBLE PULSE GENERATOR
TRIGGER
FROM PEAK COMPARATOR

FIG. 3

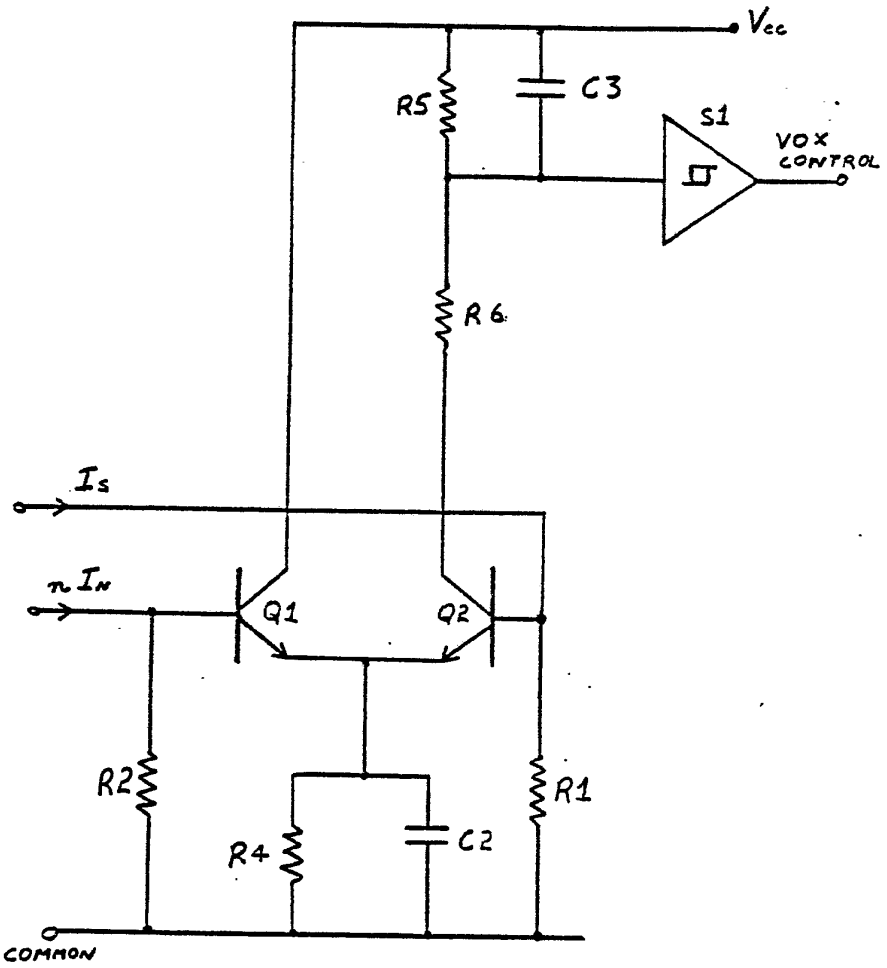


FIG. 4

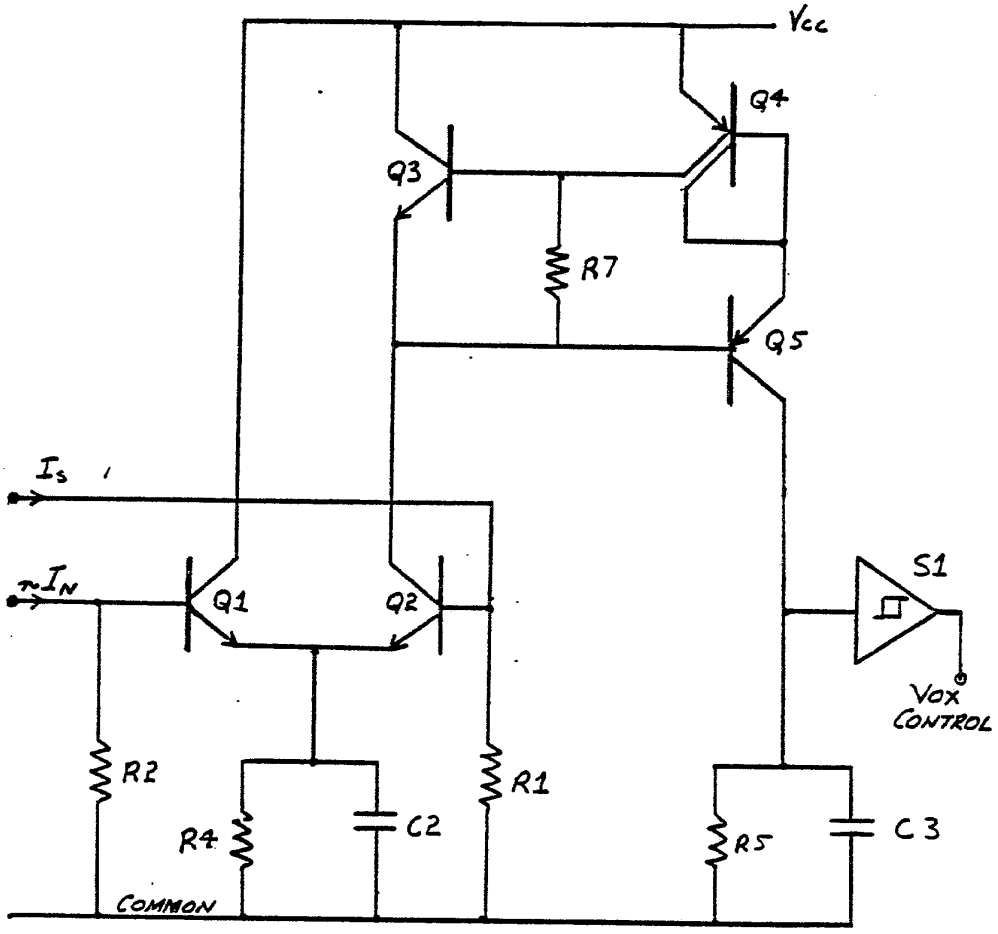


FIG. 5

INTERNATIONAL SEARCH REPORT

International Application No PCT/AU 85/00121

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [6]

According to International Patent Classification (IPC) or to both National Classification and IPC

Int. Cl.[4] G10L 5/00, H04B 7/00, 1/44, 1/46

II. FIELDS SEARCHED

Minimum Documentation Searched [7]

Classification System	Classification Symbols
IPC	G10L 5/00, 7/08, H04B 7/00, 1/44, 1/46

Documentation Searched other than Minimum Documentation to the Extent that such Documents are included in the Fields Searched [8]

AU: IPC as above; Australian Classification 01.7, 05.5

III DOCUMENTS CONSIDERED TO BE RELEVANT [9]

Category [10]	Citation of Document, [11] with indication, where appropriate, of the relevant passages [12]	Relevant to Claim No. [13]
Y,P	WO,A, 85/01411 (KELLER, PEIKER) 28 March 1985 (28.03.85)	1-3
Y	DE,A, 2931604 (SIEMENS AG) 5 February 1981 (05.02.81)	1
A	DE,A, 2849938 (KIEPE ELECTRIC GMBH) 29 May 1980 (29.05.80)	
Y	FR,A, 2305909 (ELECTRONIQUE MARCEL DASSAULT) 22 October 1976 (22.10.76)	1-3
Y	GB,A, 1516100 (SECRETARY OF STATE FOR DEFENCE) 28 June 1978 (28.06.78)	1
A	GB,A, 2001220 (ELLER) 24 January 1979 (24.01.79)	
A	US,A, 4215241 (PINKNEY) 29 July 1980 (29.07.80)	
A	US,A, 3814856 (DUGAN) 4 June 1974 (04.06.74)	
A	US,A, 3962553 (LINDER) 8 June 1976 (08.06.76)	
A	US,A, 4484344 (GARLAND, CAMPBELL) 20 November 1984 (20.11.84)	
A	JP,A, 58-156236 (MATSUSHITA DENKI SANGYO K.K.) 17 September 1983 (17.09.83) (JAPATIC English Language Abstract)	

* Special categories of cited documents: [10]

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"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
28 August 1985 (28.08.85)	(09.09.85) 9 SEPTEMBER 1985
International Searching Authority	**Signature of Authorized Officer**
AUSTRALIAN PATENT OFFICE	R. TOLHURST

Form PCT/ISA/210 (second sheet) (January 1985)

ANNEX TO THE INTERNATIONAL SEARCH REPORT ON
INTERNATIONAL APPLICATION NO. PCT/AU 85/00121

This Annex lists the known "A" publication level patent family members relating to the patent documents cited in the above-mentioned international search report. The Australian Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

Patent Document Cited in Search Report	Patent Family Members
WO 8501411	DE 3424531
GB 2001220	DE 2731971, FR 2397750, JP 54030817
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END OF ANNEX

(19) Europäisches Patentamt
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(12) EUROPEAN PATENT APPLICATION

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(54) Method of obtaining a characteristic, and hearing instrument

(57) According to the invention, a hearing instrument comprising at least one inner microphone (6) operable to determine a sensing signal representative of an acoustic signal at a position in front of the user's eardrum - which may be an acoustic signal at a position between an ITE (or ITC or CIC) hearing instrument and the eardrum or between an earpiece and the eardrum - is used. In accordance with the invention, in front of the eardrum an acoustic signal is produced. The inner microphone creates a sensing signal representative of the acoustic signal, and the signal processing unit of the hearing instrument determines a characteristic of the user's ear canal based thereon and memorizes values indicative of the characteristic. According to a preferred embodiment, the characteristic is an acoustic coupling transfer characteristic, which is determined based on a comparison of a signal representative of the input signal of the hearing instrument's receiver and the sensing signal. In contrast to the state of the art, no control loop is necessary to adapt the gain to the ear canal characteristic, but the memorized values may be used therefor.

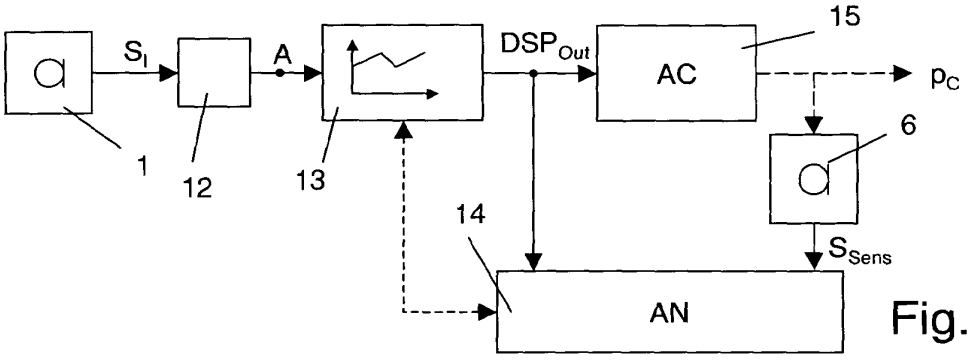


Fig. 3

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Description

FIELD OF THE INVENTION

[0001] The invention is in the field of hearing instruments. It more particularly relates to a method of obtaining a characteristic of acoustical circumstances in an ear canal of a user, to a hearing instrument, and to a method of fabricating a hearing instrument.

BACKGROUND OF THE INVENTION

[0002] The acoustical output of a hearing instrument as perceived by the user depends on the environment of the hearing instrument, especially on the ear canal properties. Conventionally, for modeling the effective gain provided by a hearing instrument placed in an ear canal, measurements in a so-called "2cc coupler" are used. However, this model system merely provides an influence of an average ear canal on the effective gain provided by a hearing instrument. The accuracy of such a model system is limited. The difference between the signal level in the real ear and the level in the 2cc coupler is often called "Real Ear to Coupler Difference" RECD.

[0003] As a consequence, when hearing instruments are set up in accordance with the needs of a user, the problem of the uncertain RECD given the individual ear canal parameters arises. The different ear canal geometries and ear canal textures of individuals' ear canals, conchas and neighboring tissues and organs (leading to individual ear impedances) as well as uncertainties concerning the vent geometry, leakage, hook damping and individual tubing give rise to different signal transfer characteristic even if the gain of the hearing instruments is the same.

[0004] The state of the art contains different approaches for addressing this problem. Swiss patent 678'692 teaches to place a microphone in the user's ear canal in order to measure acoustical properties. The results are analyzed by a specific device (audiometer). According to the teachings of US patent 4,596,902 and US patent 6,658,122 the hearing device comprises a microphone operable to measure the acoustic pressure in the ear canal and a control circuit for real-time adjustment of the gain characteristic if the measured acoustic pressure does not correspond to a reference acoustic pressure. Such a feedback control, however, uses up a lot of calculation power.

SUMMARY OF THE INVENTION

[0005] It is an object of the invention to provide a method of operating a hearing instrument, a hearing instrument, and a method of manufacturing a hearing instrument which address the problem of uncertain coupling of the hearing instrument to the individual ear of a user and which overcome drawbacks of prior art approaches and especially do not require too much calculation power to be used.

[0006] According to the invention, a hearing instrument comprising at least one microphone operable to determine a sensing signal representative of an acoustic signal at a place in front of the user's eardrum is used. In the case of a hearing instrument including multiple components (such as a hearing instrument comprising an in-the-ear (ITE) component (earpiece) and a behind-the-ear (BTE) component), said microphone may be physically located in the innermost component (earpiece) or may be located elsewhere and connected to the place the signal is collected from. This microphone in this text - independent of its physical location - is called "inner microphone" in contrast to the at least one "outer microphone" which converts the acoustic signal incident on the hearing instrument (or a component thereof) from the outside into an electric signal to be transformed in an output signal. In cases where the inner microphone is not placed at the position from where the acoustic signal is collected, it is connected to the position by sound conducting means. Such means may comprise a sound conducting tube, possibly including cerumen protection and/or a specifically adapted end geometry for ideal sound incoupling. As an alternative, the means may comprise a channel within a housing or other sound conductors.

[0007] The acoustic signal may for example be a signal in a closed or not closed volume between an ITE (or ITC or CIC) hearing instrument and the eardrum or a volume between an earpiece of a hearing instrument and the eardrum.

[0008] In accordance with the invention, in front of the eardrum, an acoustic signal is produced. The inner microphone creates a sensing signal representative of the acoustic signal, and the signal processing unit of the hearing instrument determines a characteristic of acoustical circumstances in the user's ear canal based thereon and memorizes values indicative of the characteristic.

[0009] In this text, a signal processing unit may comprise a single digital signal processor (DSP) possibly including analog-to-digital (A/D) and digital-to-analog (D/A) conversion stages. An analog amplifier may be also integrated or may be provided separately. This includes, as an example, a class D amplifier that is directly fed by a pulse width modulated signal from the signal processing unit and which redundantizes a classical D/A converter. As an alternative, the signal processing unit may comprise two or more communicatively coupled entities, for example a digital signal processor and physically separate A/D and/or D/A converters. It may also comprise a plurality of processors and/or

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other digital and/or analog signal processing elements. The signal processing unit also comprises a data memory, which is for example integrated in the digital signal processor(s) or in another element or which may be provided as separate data memory. Different elements constituting the signal processing unit do not need to be physically grouped together but may even be distributed between different hearing instrument components.

[0010] According to the invention, the hearing instrument's signal processing unit may calculate a characteristic of the ear canal which is used for optimized signal processing, for example for adapting a gain to the individual properties of the ear canal (vent leakage, impedance, etc.). Therefore, on the one hand, a separate device for measuring ear canal characteristic is not necessary, and the user of the hearing instrument does not have to go to the hearing aid professional in order to adapt the hearing instrument to the ear canal characteristic. On the other hand, the approach in accordance with the invention eliminates the need for a control loop - as has been presented in the state of the art - which would use up a lot of calculating power. Also, in contrast to the state of the art, a stored characteristic may not only be used for controlling and correcting the actually perceived signal by way of a gain adjustment but may be used for other purposes as well, including diagnostic purposes, as will be explained in the following.

[0011] In accordance with a first aspect of the invention, a transfer characteristic is evaluated by the hearing instrument, i.e. by the digital signal processing (DSP) unit. The transfer characteristic is representative of the acoustic coupling from the processed electric signal, as presented at the DSP output, to the real-ear acoustic signal.

[0012] The transfer characteristic may be stored in any way suitable to store a characteristic. Examples of values representative of a transfer function include:

1) the complex values obtained by a Fourier analysis of the transfer function.

2) only the absolute values of the Fourier coefficients.

3) one absolute value for each one of a plurality of frequency bands, such as bark bands, bands in accordance with an other psychoacoustical scale, third octave bands, etc. Such absolute values may for example be obtained by averaging a number of coefficients in accordance with 2).

4) Coefficients of a Finite Impulse Response (FIR) filter approximation of an Infinite Impulse Response (IIR) approximation or similar.

5) impulse responses.

6) complex coefficients of a numerator/denominator polynomial.

7) etc.

[0013] The values may for example be stored in the signal processing unit as the RECD values. Storing of RECD values for signal processing has been known in state-of-the-art hearing instruments. In state-of-the-art hearing instruments, however, the RECD values are determined indirectly during a fitting process. One advantage of the first aspect of the invention, therefore, may be to simplify the fitting process.

[0014] In accordance with a preferred embodiment of the invention, a signal representative of an output of the digital signal processing stage is compared with the sensing signal in order to obtain the transfer characteristic and to obtain a gain adjustment to the real ear situation.

[0015] The digital signal processing stage (DSP) output is the signal after the signal processing steps including hearing loss correction, noise reduction steps, etc. The DSP output may but need not be the physical output of a signal processing 'chip'. Rather, often a signal processing chip comprises a digital-to-analog conversion stage or a processing stage transforming the digital signal into a pulse width modulated signal, which stage may be thought of as being arranged downstream of a DSP output.

[0016] In accordance with this embodiment, therefore, the hearing instrument comprises

at least one acoustic signal acquisition microphone (the outer microphone(s),

a signal processing unit,

and at least one receiver, the at least one acoustical signal acquisition microphone being operationally connected to a first input of the signal processing unit,

an output of the signal processing unit being operationally connected to an input of said receiver,

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the signal processing unit comprising a digital signal processing stage and a memory,

an output of said digital signal processing stage being operationally connected to said output of the signal processing unit,

the hearing instrument further comprising an in-the-ear canal acoustic signal acquisition microphone (the inner microphone), an output of which is operationally connected to a second input of said signal processing unit,

the signal processing unit being operable to apply a gain on an input signal supplied to said first input to obtain an output signal and to supply said output signal to said output,

the signal processing unit further being operable to calculate, based on a comparison of said sensing signal supplied to said second input with an output signal of said digital signal processing stage, a transfer characteristic, and to apply a gain adjustment on said gain, the gain adjustment based on said transfer characteristic,

wherein at least one of the transfer characteristic and of the gain adjustment is stored in the memory, and wherein the same transfer characteristic is used for determining a gain adjustment for a plurality of gain calculation cycles or wherein the same gain adjustment is applied to the gain for a plurality of gain calculation cycles (or sampling cycles; the time interval during which a same gain adjustment is applied corresponds to a plurality of sampling intervals).

[0017] In a simplified model of the gain relations in a hearing instrument the incident acoustic signal is subject to the following steps to yield the acoustic signal in the user's ear:

$$\text{Data acquisition} \rightarrow \text{DSP Gain} \rightarrow \text{Outcoupling}$$

[0018] The quantity measuring the gain of the data acquisition stage (which gain is governed by the location of the microphones on the hearing instrument and their physical properties) is the input sensitivity SENSIN. The gain of the outcoupling is governed by the output sensitivity of the receiver(s) and the properties of the real ear (impedance, etc.). The output sensitivity SENSOUT of a hearing instrument is usually known for the 2cc coupler situation. The difference between this known output sensitivity and the real-ear output sensitivity is known as the RECD.

[0019] Therefore, for the effective gain (Real Ear Aided Gain, REAG) the following relation holds:

$$\text{DSP GAIN} = \text{REAG} - \text{SENSIN} - \text{SENSOUT} - \text{RECD} \quad (1)$$

[0020] If "ACOUSTIC COUPLING" is defined as

$$\text{ACOUSTIC COUPLING} = \text{SENSOUT} + \text{RECD} \quad (2)$$

it follows that

$$\text{DSP GAIN} = \text{REAG} - \text{SENSIN} - \text{ACOUSTIC COUPLING} \quad (3)$$

[0021] In equation (1), the individual RECD is the only unknown quantity of the system, since the input sensitivity as well as the output sensitivity on the coupler are known. In the present invention, preferably instead of addressing the RECD, ACOUSTIC COUPLING is quantified.

[0022] In modern hearing instruments, the DSP gain is situation dependent. It is influenced by user chosen programs (for example direction sensitive or omnidirectional), environmental conditions (adaptive noise suppression, possibly for special kinds of noise), etc. Therefore, in general the relationship between an input signal and an output signal is non-linear.

[0023] It is an insight of the inventor of the present application that, however, the acoustic coupling as part of the pure electro-acoustic system is approximately linear and that it therefore may be viewed as independent of the sound level and as a consequence also independent of the DSP GAIN, which latter may vary dynamically depending on the situation. This means that the acoustic coupling may be characterized by an essentially time independent transfer function $T(f)$, where f denotes the frequency. The transfer function is independent of the DSP gain and at least approximately also of the acoustic signal produced by the receiver, since the DSP-output-to-real-ear-sound-pressure-transfer

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is at least approximately linear. The spectrum of the signal is cancelled out when the transfer function is measured. It is, therefore, preferred to compare the output of the DSP (i.e. the digital or - less preferred - possibly, if the amplification is load independent, the analog signal after the gain stage) with the sensing signal. In other words, in contrast to the situation in dynamic control circuits, where an input signal or an uncorrected output of the digital signal processor is compared with the acoustic signal in the ear canal, the approach presented here includes using for the comparison a signal which is representative of the electric input signal of the receiver. For example, the signal used for the comparison may be the receiver input signal before Digital-to-Analog conversion and amplification, i.e. a signal at least approximately proportional to the receiver input signal.

[0024] It is for this reason that this approach of using the acoustic coupling redundantizes a control circuit dynamically controlling the effective signal level in the ear. It suffices to obtain a transfer characteristic once or repeatedly at intervals which are long compared to the sampling interval and to characteristic sound signal variation time constants. For example in the case of digital signal processing, the rate with which a transfer characteristic is obtained (if the transfer characteristic is obtained regularly at all) is at least 1000, preferably at least 1'000'000 times lower than the signal processing unit's sampling rate. The transfer characteristic may for example be obtained only during the fitting process and/or once every day (when the hearing instrument is switched on), once every hour, upon incidence of certain events (for example initiation by the user, battery replacement, etc.), or the like. In case of regular updates of the transfer characteristic, the interval between updates is for example always greater than 1 s, preferably greater than 1 min. or even greater than 1 h.

[0025] The transfer characteristic may, as mentioned above, for example be memorized by way of storing parameters of a transfer function $T(f)$. Alternately, instead of a transfer function, gain correction parameters may be directly stored, so that no explicit calculation of the transfer function $T(f)$ is necessary. The art provides yet further alternatives of characterizing a transfer between a (digital) electric signal and a real-ear acoustic signal, which further alternatives may also be used in accordance with the invention.

[0026] Since the transfer from the DSP output to the signal in the ear canal (the acoustic coupling) is linear, it may be characterized by comparing a DSP output representing any acoustic signal with the corresponding real ear acoustic signal. In order to determine the transfer characteristic, an arbitrary acoustic signal may be incident on the hearing instrument at a measuring time. It is not necessary to use a particular signal with a particular frequency characteristic for this. According to an embodiment of the invention, however, the arbitrary input signal may be replaced or supplemented by a special processor generated measuring signal supplied to the receiver. By this, the coherence and thus the quality of the measurement may be improved. The measuring signals may be on a non-audible level and may provide the desired functionality nevertheless due to appropriately longer averaging. Possible suited measuring signals are for example Maximum Length Sequence (MLS) signals, which as such are known in the art and are not described any further here.

[0027] The transfer characteristic may be used for adjusting a gain of the hearing instrument, which gain is then specifically adapted to the individual circumstances. Nevertheless, in contrast to control circuit approaches such as the approaches disclosed in US 4,596,902 and US 6,658,122, the adaptation to the individual circumstances does not necessarily cause a lot of computing power to be used. Rather, the same correction parameters may be used during a long period and in largely different circumstances, nevertheless yielding a good and appropriate correction. Also, in contrast to other prior art approaches, the presence of a hearing aid professional and of special equipment is not necessary in order to make this specific adaptation.

[0028] Most hearing instruments nowadays comprise at least two input microphones ("outer microphones" in this text) in order to enable beamforming. As a consequence, the hearing instruments also comprise an equal number of Analog-to-Digital (A/D) converters. According to a preferred embodiment of the invention, the inner microphone does not necessitate a further A/D converter. Rather, the hearing instrument comprises a switch by which alternatively the output of one outer microphone and of the inner microphone may be connected to the input of one of the A/D converters. This makes possible, that for occasional measurements of the transfer characteristic, the beamforming functionality is at least partially interrupted, and the characteristic may be measured without interrupting the hearing aid operation and without additional hardware.

[0029] A similar approach may be chosen for hearing instruments with only one outer microphone. In such hearing instruments, a single A/D converter input may be alternatively switched between the outer and the inner microphone. For measuring the transfer characteristic, a processor generated signal is used, and the hearing aid operation has to be interrupted. However, this procedure for example only has to be done initially when the hearing instrument is positioned at is operating place and is therefore not even necessarily perceived by the user.

[0030] The transfer characteristic may, in addition to influencing the gain characteristic of the signal processing unit or as an alternative thereto, also be used for other purposes.

[0031] As a first additional or alternative use, the transfer characteristic may be used for estimating the ear canal transfer function. Even though often in the literature no difference is made between different positions of a microphone placed in the ear canal (of an "inner microphone") or of an end of a sound conducting means leading thereto, in practice

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the distance between the microphone and the eardrum is important. In order to address this, a testing probe may be placed close (closer than 5 mm) to the eardrum and used to measure the sound level there. However, this may be done by a hearing aid professional only (due to the danger of damaging the eardrum), brings about additional efforts, and is imprecise due to the influence of the testing probe's finite size on the acoustical circumstances. However, investigations by the inventor have shown that there is a strong relation between the acoustical impedances and the transfer within the ear canal. Therefore, since the acoustic coupling contains the information on the acoustical impedances, the ear canal transfer may be estimated based on the measured acoustic coupling. This may be done using any estimation method known in the art. It may for example be based on electro-acoustical models, statistic models, discrimination or decision tree based, neuronal networks, fuzzy logic, etc.

[0032] A second additional or alternative use is an estimate of the ear impedance. The ear impedance is a basis for hearing instrument optimization and for modeling in general. In principle, ear impedance measurement requires a laborious calibration of the system and measurement of the complex acoustic coupling transfer function. The corresponding 1-microphone theory has been published in EP 1 316 783. The content of this publication is explicitly referred to, especially concerning the teaching in paragraphs [0012] to [0025] and [0027] to [0042] of the A2 publication.

[0033] A third additional or alternative use is an automatic on/off switch based on the measured acoustic coupling. This use is based on the insight that the acoustic coupling is characteristic of the situation where the hearing instrument is in place in or at the user's ear. As soon as the hearing instrument is removed, the acoustic coupling changes drastically, in which case the hearing instrument may switch off or switch to a standby-mode automatically. For this use, the acoustic coupling has to be measured repeatedly and permanently. This, however, does not imply the measurement to be real-time. A measurement once every split second, second, or plurality of seconds or even minutes may be sufficient.

[0034] As a fourth additional or alternative use, the acoustic coupling is used for making a diagnosis. This may be a diagnosis on the hearing instrument hard- or software or on the middle ear. For example, by means of characterizing the acoustic coupling, a clogging of the vent, the microphone or of the receiver outlet may be detected.

[0035] In accordance with a second aspect of the invention, the inner microphone of a hearing instrument according to the invention may also be used to measure other transfer functions than the transfer characteristic of the path from the digital signal to the acoustic signal in the ear canal.

[0036] A first example of such other characteristic is the "real ear occluded gain" transfer function. In addition to the signal provided by the hearing aid receiver(s), the user also perceives the acoustic signal bypassing the hearing instrument, through vent and guided by the human tissue, the instrument casing, etc. The output-input-relationship of this bypassing signal is called the real ear occluded gain (REOG). When the REOG is measured, the electro-acoustical amplification path of the hearing instrument is switched off, and the stimulating signal corresponds to the signal incident from outside, potentially supported by an additional source.

[0037] A second example of such other characteristic is the measurement of the acoustical feedback limit, i.e. the maximum achievable acoustical amplification for compensating the hearing loss. The acoustical feedback limit is not identical to the limit determined by the maximal DSP gain and standard RECD. It rather corresponds to the individual maximal acoustic gain. The acoustical feedback limit is not primarily dependent on the acoustic coupling, but depends on the path vent (+ housing + tissue) - outcoupling - transfer to the outer microphone. This transfer is covered if the transfer from the inner microphone to the outer microphone is measured. Measurement of the acoustical feedback limit in accordance with this aspect of the invention, therefore, comprises supplying the receiver with a processor generated signal and comparing the sensing signal obtained by the inner microphone with the signal produced by at least one outer microphone upon incidence of the signal transmitted from the receiver back via vent, housing and human tissue.

[0038] The term "hearing instrument" or "hearing device", as understood in this text, denotes on the one hand hearing aid devices that are therapeutic devices improving the hearing ability of individuals, primarily according to diagnostic results. Such hearing aid devices may be Behind-The-Ear (BTE) hearing aid devices or In-The-Ear (ITE) hearing aid devices (including the so called In-The-Canal (ITC) and Completely-In-The-Canal (CIC) hearing aid devices, as well as partially and fully implanted hearing aid devices). On the other hand, the term stands for devices which may improve the hearing of individuals with normal hearing, e.g. in specific acoustical situations as in a very noisy environment or in concert halls, or which may even be used in the context of remote communication or of audio listening, for instance as provided by headphones.

[0039] The hearing devices addressed by the present invention are so-called active hearing devices which comprise at the input side at least one acoustical to electrical converter, such as a microphone, at the output side at least one electrical to acoustical converter, such as a loudspeaker (often also termed "receiver"), and which further comprise a signal processing unit for processing signals according to the output signals of the acoustical to electrical converter and for generating output signals to the electrical input of the electrical to mechanical output converter. In general, the signal processing circuit may be an analog, digital or hybrid analog-digital circuit, and may be implemented with discrete electronic components, integrated circuits, or a combination of both. In the context of this application, signal processing

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units comprising digital signal processing means are preferred.

BRIEF DESCRIPTION OF THE DRAWINGS

[0040] In the following, embodiments of the invention are described with reference to drawings. The drawings are all schematical and show:

- Fig. 1 diagram of a hearing instrument according to the invention,
- Fig. 2 an ear canal with an in-the-ear-canal component of a hearing instrument according to the invention,
- Fig. 3 a diagram illustrating processing steps of a method according to the invention,
- Fig. 4 a diagram illustrating the transfer steps of the acoustic coupling,
- Fig. 5 graphs illustrating an example of a gain correction,
- Fig. 6 a diagram illustrating processing steps of an alternative embodiment of the method according to the invention,
- Fig. 7 a diagram illustrating the use of an A/D converter of a hearing instrument comprising multiple outer microphones,
- Fig. 8 a diagram illustrating the measurement of the REOG,
- Fig. 9 a diagram illustrating the measurement of the acoustic feedback limit,
- Fig. 10 a diagram illustrating the estimation of the ear canal transfer function,
- Fig. 11 a diagram illustrating further embodiments of the method according to the invention.

[0041] In the figures, corresponding components are provided with the same reference numerals.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0042] The hearing instrument of **Figure 1** comprises at least one outer acoustic-to-electric converter (microphone) 1 (often, two or even three acoustic-to-electric converters are available in each hearing instrument), a signal processing unit (SPU) 3 operable to apply a time- and/or frequency-dependent gain to the input signal or input signals S_I resulting in an output signal So and at least one electric-to-acoustic converter (receiver) 5. The hearing instrument further comprises an inner acoustic-to-electric converter 6.

[0043] In **Figure 2**, very schematically an ear canal 8 with an inserted in-the-ear-canal component, namely an otoplastic 7, of a behind-the-ear hearing instrument is illustrated. In general, an in-the-ear-canal component or an in-the-ear component of a hearing instrument in the context of this application is also called earpiece. The sound output signal is guided from the receiver to an interior of the ear canal 8 by means of a sound conducting tube 9 held by the otoplastic 7. The otoplastic 7 is an element shaped to fit in the user's ear, which comprises next to holding means for holding the sound conducting tube also a vent 2 for pressure equalization. The vent, and, more generally, the earpiece or the hearing instrument may have any shape as such know in the art including large vents (such as IROS venting), or limited open fittings (with or without otoplastics).

[0044] The inner microphone 6 is integrated in the otoplastic. In the figure, the eardrum is denoted by 10, the volume between the eardrum and the otoplastic by 11. In contrast to the shown embodiment, the inner microphone may be placed in an interior of the earpiece (or the CIC, ITC, or ITE hearing instrument) and may be connected to the volume 11 by a channel. It may also be arranged adjacent to the sound conduction means from the receiver to the volume or adjacent to the vent.

[0045] In the shown embodiment, a digital signal processor (a core part of the signal processing unit) and preferably the entire signal processing unit is placed in the behind-the-ear component (not shown), together with the outer microphone(s), the receiver, and a battery compartment. The hearing instrument comprises a sensing signal transmission wire 4 connecting the inner microphone with the signal processing unit. Signal transmission between the inner microphone and the signal processing unit may as an alternative be wireless, in which case, however, the in-the-ear-canal component has to comprise an energy source.

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[0046] In the shown embodiment of a BTE hearing instrument, the inner microphone is placed in the earpiece, whereas the outer microphone and the receiver are arranged in the behind-the-ear component. Thus there is a sound conducting connection (sound conducting tube 9) between the receiver and the earpiece, and an electrical signal conducting connection between the inner microphone and the behind-the-ear-component. However, it is also possible to either place the inner microphone in the behind-the-ear component, in which case there is a sound conducting connection between the earpiece and the inner microphone, or to place at least one receiver in the earpiece (in which case there is an electrical signal conducting connection between the behind-the-ear component and the receiver), or both. It is even possible to place the outer microphone in the earpiece, although this alternative is clearly less preferred.

[0047] Instead of a behind-the-ear (BTE) hearing instrument, the instrument may also be an in-the-ear (ITE) or in-the-canal (ITC) hearing instrument, including a completely-in-the-canal (CIC) hearing instrument. In such a hearing instrument, the outer microphone(s) is/are for example placed on an outside facing side of the instrument, whereas the inner microphone is placed on the inside facing the eardrum or is connected to the inside by sound conducting means. In an ITE, an ITC or a CIC hearing instrument, the in-the-ear component is often the only constituent of the hearing instrument, i.e. the hearing instrument may consist of its in-the-ear-canal component. As yet another alternative, the in-the-ear component may be an in-the-ear-canal component of the kind described in the European patent application 05 405 022.4.

[0048] A very schematic diagram of the signal path in a basic embodiment of the invention is shown in **Figure 3.** The outer microphone 1 produces an (analog) electric signal which is the input signal S_I for the signal processing unit. The signal processing unit comprises a signal collecting stage 12. The signal collecting stage 12 preferably includes an analog-to-digital converter and may include further functionality such as analysis, the calculation of noise reduction parameters, etc. The signal processing unit further comprises a signal processing stage 13, where the input signal is transferred into a (digital) output signal. This may include having a frequency dependent gain act upon the signal, which frequency dependent gain may depend on parameters such as acoustic signal direction of incidence, signal level, detected noise levels, etc. Signal collecting stage 12 and signal processing stage 13 may together have the functionality of any known or yet to be developed signal management in hearing instrument technology.

[0049] The frequency dependent gain may, according to a preferred embodiment of the invention, be further dependent on a transfer characteristic, as will be explained in the following.

[0050] The output signal DSP_{Out} of the signal processing stage 13 is converted into an analog electric signal, amplified and converted into an acoustic signal. These steps are illustrated in **Figure 4**, where 21 denotes a digital-to-analog converter, 22 an amplifier, 5 a receiver and 23 the sound conduction by tubing (if, for example, the hearing instrument is of the behind-the-ear-type) and the influence of the earmold (otoplastic).

[0051] Returning to Figure 3, conversion of the DSP_{Out} signal into the real-ear acoustic signal in front of the eardrum is denoted by a single step "acoustic coupling" in the figure. As explained, for the purpose of the first aspect of the invention, it is sufficient to measure the acoustic coupling as a whole. Knowledge of the mechanisms underlying the individual steps - as represented in Fig. 4 - is not necessary. Note that in the representation of Fig. 3, as well as in the following description, the acoustic coupling 15 includes the digital-to-analog conversion and the amplification by the analog amplifier. The term "acoustic coupling", therefore, in this description is used for the sum of steps leading to the conversion of the processed digital output signal into the real-ear acoustic signal in the ear canal.

[0052] The ear canal sound pressure level pc in the volume 11 in front of the eardrum is sensed by the inner microphone 6. The sensing signal S_{sens} output by the inner microphone 6 is compared to a signal DSP_{Out} representative of the receiver input by an analyzer 14. The transfer characteristic may for example be represented by explicitly known parameters of a transfer function $T(f)$ or as an alternative by appropriate gain correction values, etc. The transfer characteristic is supplied to the signal processing stage and preferably has an influence on the effective gain values. The DSP may control the analyzer 14; it may for example trigger a measurement, define the measurement parameter, etc. For example, if the transfer characteristic reveals that the acoustic signal in a particular frequency region is suppressed stronger than average, the gain calculated by the signal processing unit based on the input signal and prestored information is corrected by a corresponding increase in said frequency region. A simplified example of an evaluation of a gain correction $C(f)$ is very schematically shown in **Figure 5**. The acoustic coupling transfer function $T(f)$ of a signal 51 is compared with an average transfer function 52 which may have been obtained as an average of a large number of measurements or by a measurement with a 2cc coupler, which is factory stored in the signal processing unit and to which the uncorrected gain calculation is adapted. From the difference of the signal 51 and the average signal 52, a gain correction $C(f)$ is evaluated. The gain correction may be stored in the signal processing unit and be applied to the gains evaluated thereby during operation of the hearing instrument. Since the acoustic coupling is linear and the acoustic coupling transfer function essentially time and acoustic signal independent, so is the gain correction. Therefore, applying the once evaluated gain correction C to the input signal a plurality of times always results in an appropriately corrected gain. The dots 54 in the right panel of figure 5 illustrate a discretized version of the gain correction for the case the gain is evaluated discretely in a number of frequency bands. Applying the gain correction may then just be an addition of the correction values C_f to the calculated gain values. The correction values C_f are indicative

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of the transfer characteristic, and storing a number of discrete gain correction values C_f is also a preferred way of storing the characteristic in the signal processing unit.

[0053] According to a further, less preferred embodiment, instead of a signal representative of the receiver input, a signal of a different stage in the signal processing may be used, for example an input signal of the signal processing stage 13 (tapped at point A in the figure).

[0054] As mentioned, it has been found that the acoustic coupling is approximately linear and that it therefore may be viewed as independent of the sound level and as a consequence also independent of the operations of the signal collecting stage 12 and of the signal processing stage 13. This means that the acoustic coupling may be characterized by a transfer function $T(f)$: $S_{sens}(t, f) = T(f)DSP_{out}(t, f)$, where f denotes the frequency and t the time. Therefore, the analyzer can characterize the acoustic coupling once (or repeatedly with a repetition rate that is small compared to the sampling rate of the digital signal processor, for example during the fitting process and once every hour, or once every day), and the characterization parameters - for example parameters of a transfer function - may be stored. The characteristic may in the following be used for setting an appropriate, situation adapted gain or the like.

[0055] **Figure 6** shows an alternative embodiment of a hearing instrument. The illustrated hearing instrument is distinct from the hearing instrument of Fig. 3 in that it comprises a signal generator 17 for generating a measuring signal in order to potentially enhance the quality of the measurement. The measuring signal may be admixed to the processed input signal, or it may be used instead of the latter. The corresponding adding stage 18 and switch 19 are also illustrated in the figure.

[0056] The hearing instrument of **Figure 7** is of the type comprising at least two outer microphones in order to enable beamforming. Each outer microphone is allocated an Analog-to-Digital converter 31.1, 31.2 (which may be integrated in a signal processing unit and which is comprised in the signal collecting stage 12 in the above figures; if the A/D converters are not integrated, instead of two separate A/D converters, in practice often a dual A/D converter comprising two inputs and two outputs will be used). Also the inner microphone output signal, of course, has to be analog-to-digital-converted in the case of digital signal processing. Corresponding analog-to-digital converters are not shown separately in Fig. 3 and 6, but are assumed to be integrated in the analyzer 14. However, due to the approach in accordance with the invention, the inner microphone 6 output signal does not permanently need to be processed and as a consequence does not permanently require an analog-to-digital converter. The hearing instrument of Figure 7 comprises a switch 32 by which alternatively the output of one outer microphone 1.2 and of the inner microphone 6 may be connected to the input of one of the A/D converters. This makes possible, that for occasional measurements of the transfer characteristic, beamforming functionality is at least partially interrupted (if the hearing instrument is not anyway in the "omni" mode in which only one microphone signal is required), and the characteristic may be measured without interrupting the hearing aid operation and with the switch 32 as only additional hardware. In the representation of Figure 7, the functionality of the analyzer is shown as integrated in the digital signal processor 33.

[0057] Figures 8-11 illustrate different operation modes of the hearing instrument of either of the hearing instruments sketched in the previous Figures. In accordance with **Figure 8**, a real ear occluded gain (REOG) characteristic is measured. For this measurement, the amplification path of the hearing instrument is switched off, and merely the output signals of the outer microphone 1 and of the inner microphone 6 are compared to provide a REOG transfer characteristic. This may be by way of a REOG transfer function $T_{REOG}(f)$, where for example $S_{SENS}(f,t) = T_{REOG}(f)^* S_I(f,t)$.

[0058] A set-up for measuring the real-ear acoustical feedback limit is shown in **Figure 9**. A measuring signal generated by the signal generator 17 is output via the receiver, preferably in a situation, where no or little external noise is present. The analyzer compares the sensing signal S_{Sens} by the inner microphone with the signal S_I detected by the outer microphone. From this comparison, the analyzer can compute the individual maximum gain which information is used by the signal processing stage 13 in subsequent signal processing.

[0059] **Figure 10** illustrates the estimation of the ear canal transfer characteristic. By this estimation, the difference between the real-ear sound level at the place of the inner microphone and at the position of the eardrum is addressed. The ear canal transfer 41 is illustrated by a corresponding box in the figure. The ear canal transfer characteristic estimator 42 is provided with appropriate means of estimating the ear canal transfer characteristic from the acoustic coupling transfer characteristic.

[0060] Further uses of the knowledge of the acoustic coupling transfer characteristic are summarized in **Figure 11**. According to one embodiment, the additional processing stage 61 serves for calculating or estimating the ear impedance (IM). According to another embodiment, the additional processing stage 61 determines whether or not the hearing instrument is properly worn by the user. This allows the hearing instrument to control a corresponding smart on/off-switch by which for example the consumption of electricity may be drastically reduced in case the hearing instrument is not worn. According to yet another embodiment, the further processing stage calculates a quantity (DIAG) which may be used for diagnosing a status of either the person wearing the hearing instrument or the hearing instrument itself or both. The uses of Figures 10 and 11 may, of course, be combined.

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Claims

1. A method of obtaining a characteristic of acoustical circumstances in an ear canal of a user, in which a hearing instrument or a hearing instrument component is placed, the hearing instrument including at least one outer microphone (1, 1.1, 1.2), a signal processing unit (3), at least one receiver (5), and at least one inner microphone (6) operable to obtain a sensing signal from an acoustic signal at a position in front of the user's eardrum (10), the method comprising the steps of producing an acoustic signal in at least a part of the ear canal, of obtaining, by means of said inner microphone, a sensing signal representative of said acoustic signal, of determining, by said signal processing unit and from said sensing signal, a characteristic of the acoustical circumstances, and of memorizing values indicative of said characteristic for further use.

2. A method according to claim 1, wherein said acoustic signal is produced by said at least one receiver, and wherein said characteristic is a transfer characteristic.

3. A method according to claim 2, wherein the acoustic signal is directed from the receiver to the ear canal by means of a sound conducting tube positioned at least partially in the ear canal.

4. A method according to claim 2 or 3, wherein signal processing of a signal obtained from the at least one outer microphone (1, 1.1, 1.2), includes a digital signal processing stage, and wherein for determining the transfer characteristic, a signal representative of an output signal of said digital signal processing stage is compared to said sensing signal.

5. A method according to any one of claims 2-4, wherein said transfer characteristic is given by a transfer function ($T(f)$), and wherein said values are parameters of the transfer function.

6. A method according to any one of claims 2-5, wherein receiver input signal is supplied by a processor generated measuring signal for generating said acoustic signal.

7. A method according to any one of claims 2-6, wherein a gain acting on an input signal S_I of the signal processing unit is adjusted based on the transfer characteristic.

8. A method according to any one of claims 2-7, wherein the transfer characteristic is used to estimate an ear canal transfer function indicative of an acoustic transfer property within the ear canal to the eardrum.

9. A method according to any one of claims 2-8, wherein the transfer characteristic is used to estimate an ear impedance.

10. A method according to any one of claims 2-9, wherein the transfer characteristic is used to determine whether the hearing instrument is properly worn by the user, and wherein the hearing instrument is switched off or is switched to a standby mode if it is found not to be worn by the user.

11. A method according to any one of claims 2-10, wherein the transfer characteristic is used to make a diagnosis on a state of the hearing instrument or on a status of the user.

12. A method according to any one of the previous claims, wherein the transfer characteristic is obtained upon a user induced event and/or is obtained repeatedly at regular or irregular intervals during operation of the hearing instrument, wherein said intervals are greater than a signal processing unit sampling interval by at least a factor of 1000.

13. A method according to claim 1, wherein the acoustic signal is produced by an acoustic source outside of the ear canal, wherein, for the determination of the sensing signal, any amplification by the hearing instrument is switched off, and wherein the characteristic is a real ear occluded gain transfer characteristic.

14. A method according to claim 1, wherein the input of the at least one receiver is supplied by a processor generated measuring signal for generating said acoustic signal, and wherein said characteristic is an acoustical feedback limit measured by comparing the sensing signal with an acoustic signal detected by at least one outer microphone.

15. A hearing instrument comprising an in-the-ear component adapted to be at least partially placed in the ear canal of a user, the hearing instrument further comprising at least one outer microphone (1), a signal processing unit (3)

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comprising a data memory, and at least one receiver (5), the signal processing unit being operable to transform an input signal provided by said at least one outer microphone into an output signal supplied to said at least one receiver, the hearing instrument further comprising at least one inner microphone (6) operable to obtain a sensing signal from an acoustic signal at a position in front of the user's eardrum (10), an output of the inner microphone being operatively connected to an input of the signal processing unit, the signal processing unit being operable to obtain from the sensing signal provided by said inner microphone, a characteristic of acoustical circumstances in the user's ear canal, and to memorize, in said data memory, values indicative of said characteristic for further use.

16. A hearing instrument according to claim 15, the output of the inner microphone not being part of a control loop.

17. A hearing instrument according to claim 15 or 16, wherein the signal processing unit comprises a digital signal processing component and an analyzer entity (14), the analyzer entity operable to carry out a comparison between said sensing signal and a an output of said digital signal processing component, to evaluate said characteristic from said comparison, and to provide said characteristic to the digital signal processing component.

18. A hearing instrument according to any one of claims 15-17, wherein the signal processing unit comprises a signal generator (17) operable to generate a processor generated measuring signal supplied to said at least one receiver (5).

19. A hearing instrument according to any one of claims 15-18, the in-the-ear component comprising a vent (2).

20. A hearing instrument according to claim 19, the in-the-ear component being an open fitting element.

21. A hearing instrument according to any one of claims 15-20 being a behind-the-ear hearing instrument comprising a behind-the-ear component, the behind-the-ear component including a digital signal processor of the signal processing unit, the in-the-ear component being an earpiece connected to the behind-the-ear component by sound conduction means and further by signal transmission means operable to transmit the sensing signal to the digital signal processor.

22. A hearing instrument according to claim 21, wherein the inner microphone is placed in the earpiece, and wherein the hearing instrument comprises an electrical wiring between the inner microphone and the digital signal processor.

23. A hearing instrument according to claim 21, wherein the inner microphone is placed in the behind-the-ear component and wherein the hearing instrument comprises a sound conduction tubing between a position fixed by the earpiece and the inner microphone.

24. A hearing instrument according to any one of claims 21-23, wherein the receiver or all receivers are placed in the behind-the-ear component, and wherein the hearing instrument comprises a sound conduction tubing between the receiver and a position fixed by the earpiece.

25. A hearing instrument according to any one of claims 21-23, wherein at least one receiver is arranged in the earpiece, and wherein the hearing instrument comprises an electrical wiring between the signal processing unit and the receiver.

26. A hearing instrument according to any one of claims 15-20 being an in-the-ear, in-the-canal or completely-in-the-canal hearing instrument.

27. A method of manufacturing a hearing instrument, the method comprising the steps of

- assembling at least one outer microphone (1), a signal processing unit (3), at least one receiver (5) and at least one inner microphone, the inner microphone forming part of an in-the-ear component or comprising sound conducting tubing connecting it to the in-the-ear component, the signal processing unit comprising a data memory,

- of establishing operational connections between an output of the at least one outer microphone and the signal processing unit, between an output of the signal processing unit and an input of the at least one receiver, and between an output of the at least one inner microphone and a further input of the signal processing unit,

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- and of providing the signal processing unit with a software enabling the signal processing unit to obtain from a sensing signal provided by said inner microphone, a characteristic of acoustical circumstances in the user's ear canal, and to memorize, in the data memory, values indicative of said characteristic for further use.

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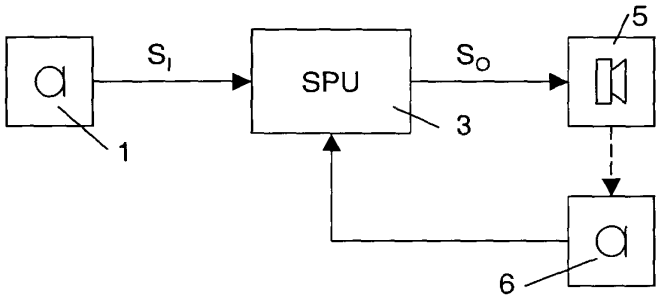


Fig. 1

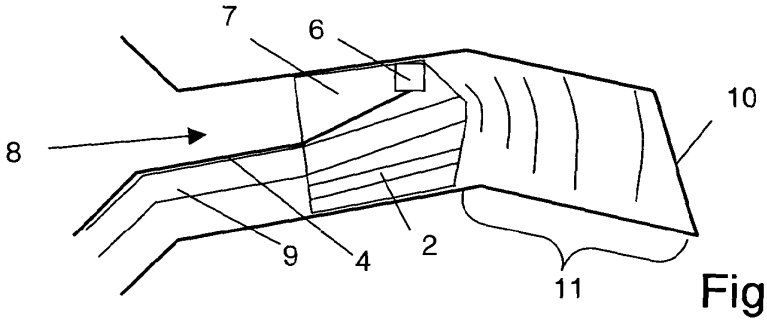


Fig. 2

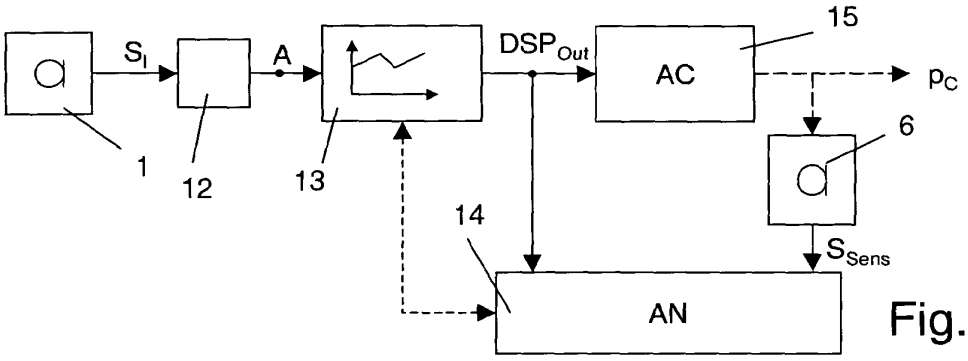


Fig. 3

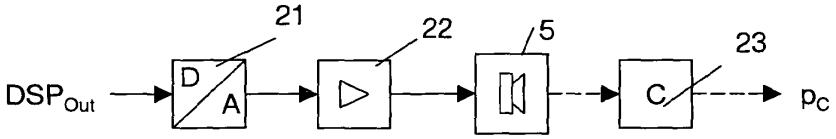


Fig. 4

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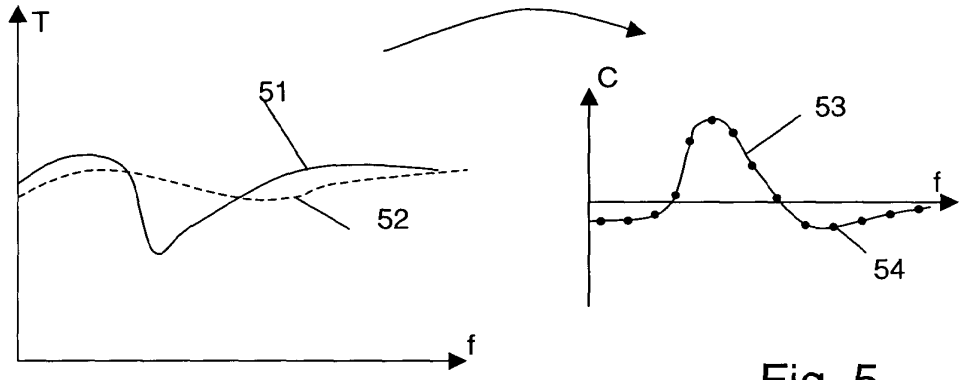


Fig. 5

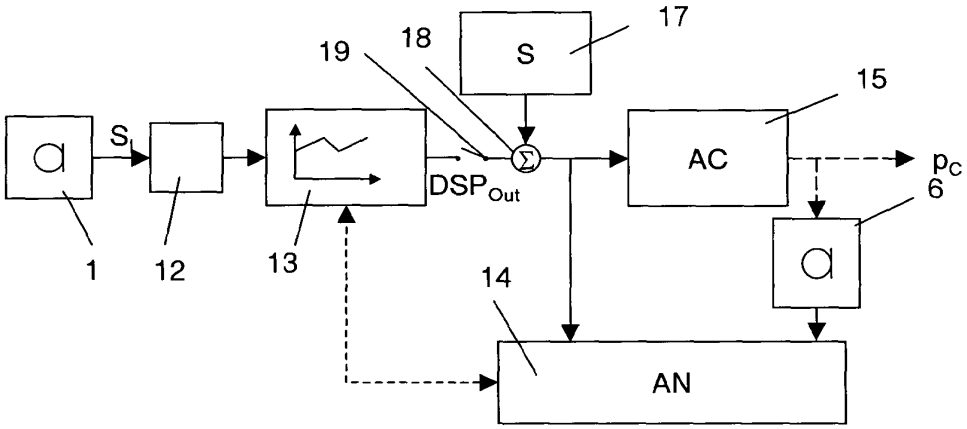


Fig. 6

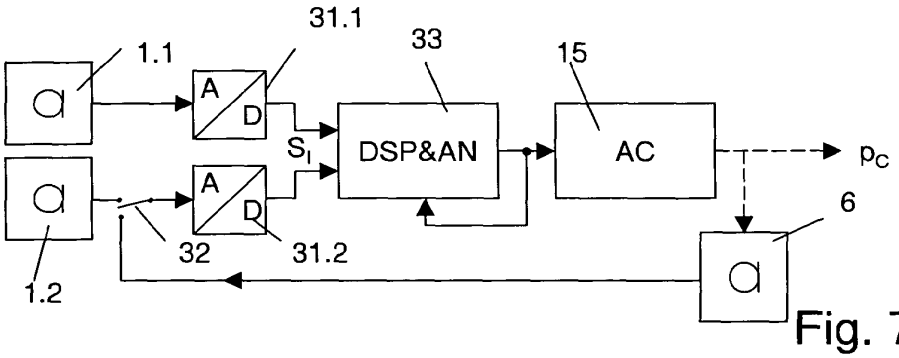


Fig. 7

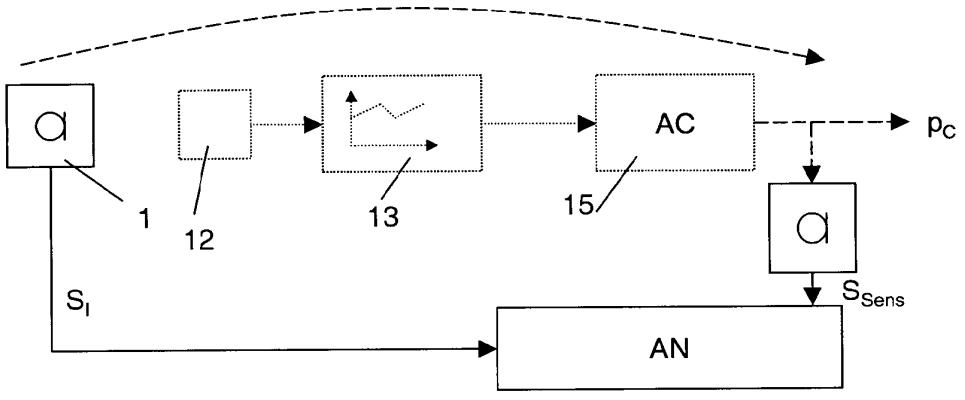


Fig. 8

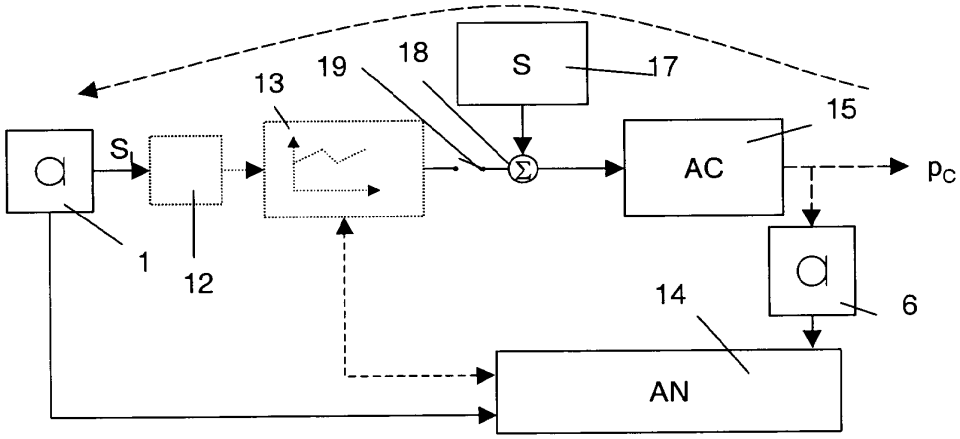


Fig. 9

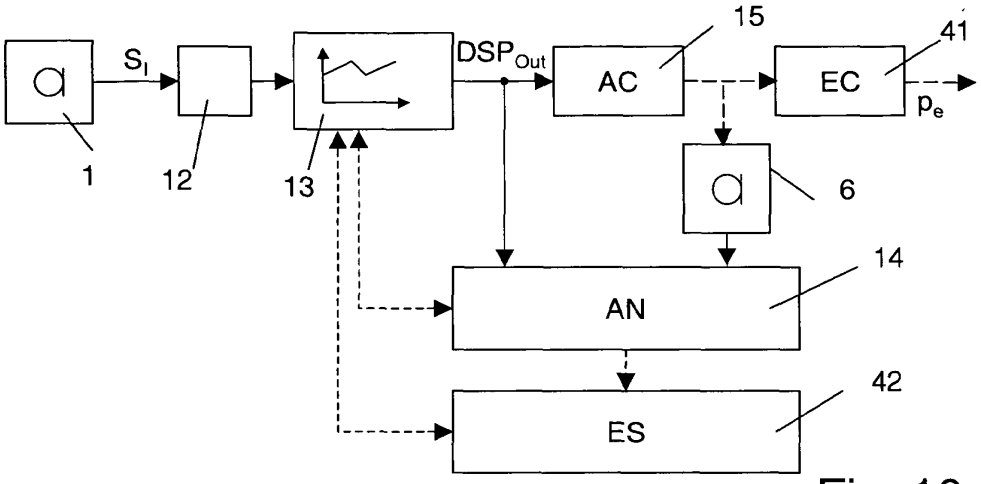

15
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DSP_{Out}
AC
EC
p_e
1
12
13
6
14
AN
42
ES

Fig. 10

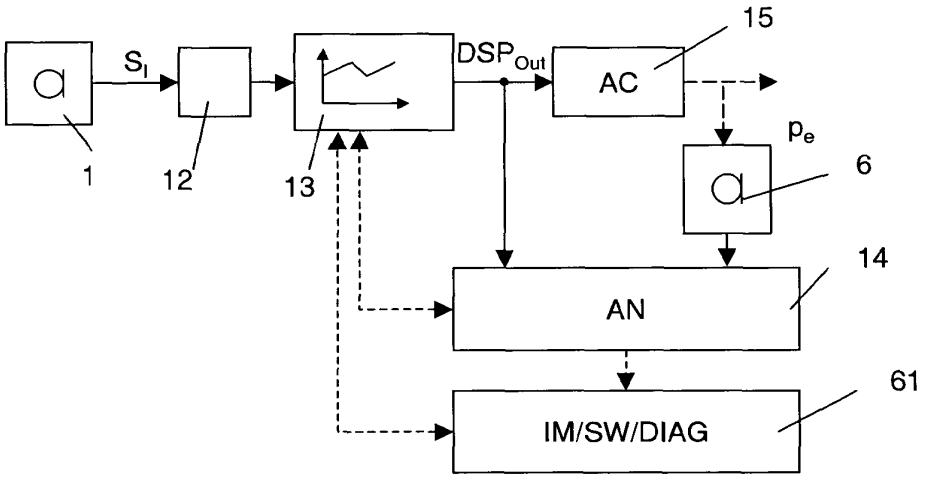

15
S_I
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61
IM/SW/DIAG

Fig. 11

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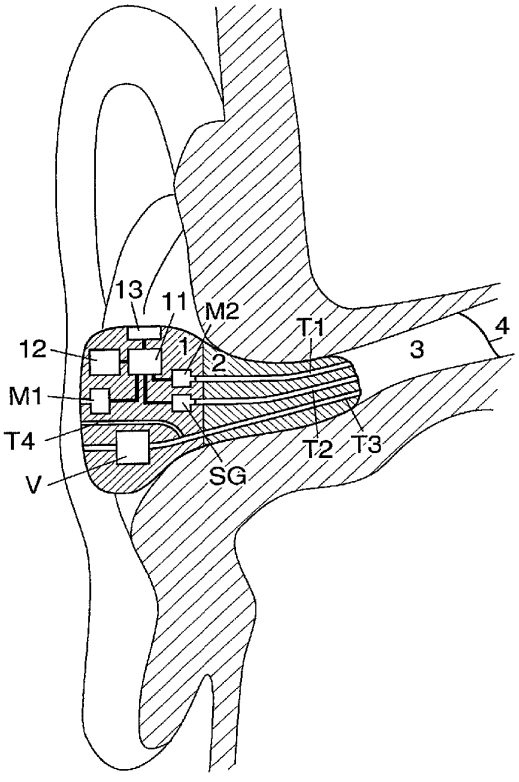
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[Continued on next page]

(54) Title: EAR TERMINAL WITH A MICROPHONE DIRECTED TOWARDS THE MEATUS



(57) Abstract: Ear terminal comprising a sealing section (2) for arrangement in the meatus (3) of a human, comprising: an inner microphone (M2) having a sound inlet (S2) for being directed, directed into the meatus, an electronics unit (11) coupled the inner microphone (M2) and also being coupled to a power supply (12) as well as an outer microphone (M1) for measuring acoustic signals in the environment.

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SK, SL, TJ, TM, TR, TT, TZ, UA, UG, US, UZ, VN, YU, ZA, ZW.

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EAR TERMINAL WITH A MICROPHONE DIRECTED TOWARDS THE MEATUS.

The invention concerns the physical design of an adaptive hearing protective earplug combined with an audio communications terminal.

There exist a lot of solutions for hearing protection and audio communication in noisy environments based on earplugs and ear-muffs with earphones (loudspeakers), boom microphones, cheek-bone microphones, or throat microphones. All these solutions have one of more of the following undesirable properties:

- heavy and clumsy.
- uncomfortable.
- inferior quality of sound pick-up and restoration.
- poor noise attenuation.
- attenuate both desired and unwanted sounds.

It is an object of this invention to provide an ear terminal having none of these shortcomings, being a lightweight, all-in-the-ear intelligent hearing protector with wireless communication. The noise attenuation is automatically adapted to the noise conditions and communication modes. The present invention therefore simultaneously protects the hearing and provides improved communication abilities in different noise environments. It is intended for continuous use during the working day or other periods when hearing protection and/or voice communication is needed.

The invention also concerns a device for utilising the speech sound produced in the ear of a person carrying hearing protective communications ear plugs according to the invention.

Present day devices intended to pick up speech from a person in a very noisy environment represent a technological challenge and take several forms. Common types include

- A microphone in close proximity to the mouth, carried on a microphone boom. The microphone is made with a characteristic emphasising the near field from the mouth. This type is sometimes referred to as "noise cancelling".
- A vibration pickup in contact with the throat, picking

up the vibrations of the vocal cord.

- A vibration pickup in contact with the wall of the meatus, the outer ear canal, picking up the vibrations of the tissue in the head.
- A similar pickup in contact with the cheek-bone.

These device types are either fairly sensitive to acoustic noise masking the speech, or certain speech sounds are poorly transmitted, especially the high frequency consonant sounds necessary for good intelligibility.

Persons exposed to high noise levels are required by health and safety regulations to wear hearing protectors. The protectors take the form either of sealing cups which enclose the ear, or ear plugs which block the ear canal. The latter type of protector is often preferred because of its small size and relatively good comfort.

Thus it is an additional object of this invention to provide an ear plug with two desirable properties:

- The cavity sealed off in the inner portion of the meatus by the ear plug is relatively free of external noise, this is the purpose of the ear plug in protecting the hearing.
- The sound field in the cavity generated by the persons own voice contains all the frequency components necessary to reconstruct the speech with good intelligibility.

The solution according to invention takes advantage of these facts. By using a microphone to pick up the acoustic sound field in the inner portion of the meatus and processing the microphone signal according to the invention, a speech signal of high quality and low noise masking is produced.

It is an additional object of this invention to provide a system for increasing the user's feeling of naturalness of the user's own voice when using a hearing protective communications terminal according to the invention.

Using ordinary earplugs or earmuffs, the user usually feels his own voice being distorted, a feature reducing the comfort of wearing hearing protectors. Ordinary hearing defenders changes the normal sound transmission path from

the mouth to the eardrums. Thus the auditory feedback from the users own voice is affected resulting in an unintended change in the speech output. A normal response is to raise one's own voice level when using headsets or earplugs.

The invention solves this problem by filtering and mixing in the user's own voice picked up by either the outer or the inner microphone at one ear and reproduce the signal at the loudspeaker in the other ear. It is also possible to reproduce the signal by the loudspeaker in the same ear, in which case feedback cancellation has to be applied. Thus the user's voice is felt more natural both with respect to frequency response and speech level. This feature will increase the level of acceptance for continuous use of hearing protectors during the whole working day. The own voice signal is added and reproduced in such a way that the noise reduction property of the hearing protector is maintained.

An additional object of this invention is to provide a programmable personal noise exposure dose meter that measures the true exposure in the user's ear and calculates the hearing damage risk.

Present day noise exposure dose meters, also called dosimeters, usually consist of a microphone and a small electronics unit that may be attached to the body or worn in a pocket. The microphone may be mounted on the electronics unit or it may be fastened to the collar or on the shoulder. ANSI S1.25 specifies dosimeters.

Present day dosimeters have several shortcomings:

- Dosimeters do not measure the noise that actually affects the hearing organ (e.g. when the user wears a hearing protector, helmet, etc.). Even when the ear is not covered, measurements may be influenced by body shielding.
- Dosimeters are susceptible to non-intentional or intentional errors, which may influence readings, such as wearers tapping or singing into dosimeter microphones or by wind-generated noise.
- Dosimeters are inaccurate if impulse or impact noise is present.

The invention solves these problems by using a microphone that measures the sound at the eardrum and employs analysis procedures that take into account both stationary and impulsive sound. When the dose meter is part of a communications terminal this includes external noises, incoming communication signal, as well as possible malfunctioning of the equipment.

It is also an object of this invention to provide a device for verifying in situ that a hearing protector is properly used.

Present day hearing protectors take the form either of sealing cups which enclose the ear, or ear plugs which blocks the ear canal. For both types, it is critically important to avoid leakage of the noise sound through or around the sealing and blocking parts of the hearing protectors.

Experience shows that several factors may compromise the sealing of a hearing protector and thereby increase the risk of hearing damage. These factors include

- Irregular surfaces which the sealing material is not able to follow properly. Examples are spctacles used with ear cups, and ear plugs used by persons with irregularly formed ear canals.
- Improper placement of the hearing protector. Experience and patience is required by the user to get a hearing protector mounted correctly. In cases where the user is wearing a helmet or cap, the hearing protector may be accidentally pushed out of position during use.
- Ageing of the materials in the sealing may reduce the resilience of the sealing and thereby allow leakage around the sealing.

The result of leakage is reduced damping of potentially harmful noise. Ideally, the leakage should be detected and remedied prior to noise exposure. The leakage may not be clearly audible. Accordingly, noise situations may comprise of intermittent or impulsive components which may damage the hearing almost instantaneously if a hearing protector should be malfunctioning or imperfect without the user's knowledge.

The invention solves these problems by an *in situ* acoustical measurement, which is analysed and reported to the user in audible form, or to external equipment by means of communication signals. The devices necessary for the measurement are an integral part of the hearing protector. Verification may be activated by the user at any time, or be continuously running when the application is critical. Optionally, verification may be activated by other persons (or devices) than the user, e.g. to verify hearing protector function before admittance to a noisy area is allowed.

The above mentioned problems are solved by the invention according to the attached claims.

The invention will be described below with reference to the accompanying drawings, the drawings illustrating the invention by way of examples.

Fig. 1 is a simplified vertical section along the central axis of the meatus of the outer ear of an erect human, with an inserted ear terminal according to one embodiment of the invention also shown in vertical section along the axis, locally coincident with the meatus axis.

Fig. 2 is an electrical wiring diagram showing the functional components and connections between electronic components in a preferred embodiment according to the invention.

Fig. 3 is an illustration of one method according to the invention, showing that spectral analysis of sound picked up in the ear is compared with spectral analysis of sound picked up by a microphone at a standard distance, e.g. of 1 metre, under otherwise quiet conditions.

Fig. 4 is an illustration of speech sound analysis and following sound source classification with filtering conducted according to the sound source classification, according to one embodiment of the invention.

Fig. 5 is an illustration of another method according to the invention, illustrating an analysis of sound picked up close to the ear being compared with an

analysis of sound picked up by a microphone arranged in the meatus.

Fig. 6 illustrates a simplified section through a human's right and left ears with ear terminals according to the invention illustrated for improved natural sound purposes.

Fig. 7 illustrates a process diagramme for one embodiment of the invention concerning noise dose metering, here illustrating an A-weighting with accumulated noise dose measurements, and also with C-weighting for peak noise value registration.

Fig. 8 illustrates another embodiment of the invention illustrating a processing scheme for online verification of hearing protector performance.

Fig. 9 illustrates an electric analogy diagram of the acoustic phenomenon on which an embodiment for online verification of hearing protector performance is based.

Description of preferred embodiments of the invention.

The physical design of an embodiment of the present invention enables the construction of a complete all-in-the-ear hearing protector and communications terminal with strong passive sound attenuation, strong active sound attenuation, high quality sound restoration, high quality sound pick-up, small size, low weight, and comfortable fit.

One embodiment of this invention is illustrated in Fig. 1, and provides the general physical design of a complete all-in-the-ear hearing protector and communications terminal, regarded as a combination of passive sealing, characteristics and placement of electro-acoustic transducers as well as acoustic filters, electric circuitry, and a ventilation system for pressure equalisation.

The ear terminal comprises an outer section 1 arranged for sitting adjacent to the outward facing portion of the sealing section 2 and a part of the inward facing portion of the outer section 1 is formed to fit the concha around the outer portion of the meatus 3.

The physical design represented by an embodiment of the invention enables some or all of the following functionalities:

- External sounds are attenuated by a combination of passive and active noise control. The passive attenuation is obtained by means of an earplug 1,2 with a sealing system 2 inserted in the outer part of the ear canal or *meatus* 3. The active noise control is achieved by using one or two microphones M1, M2 and a loudspeaker SG together with electronic circuits in an electronics unit 11 mounted in the earplug system. The algoritmes for noise control are per se known and will not be described in any detail here, but may include active noise cancelling by feedback of acoustic signals converted by at least one of said microphones (M1,M2) throught the sound generator (SG).
- Restoration of desired sounds (external sounds and signals from the communication system) at the eardrum or *tympanum* 4 is achieved by using the same microphones M1, M2, and loudspeaker SG and the electronics unit 11. Again, the algoritmes for obtaining this are per se known and will not be described in any detail here, but may include amplification of chosen frequencies converted by said the microphone (M1) and generating a corresponding acoustic signal through said sound generator (SG). The frequencies may for example be within the normal range of the human voice.

9. Ear terminal according to claim 1, comprising a sound generator (SG) arranged for being directed toward the meatus and being coupled to said electronics unit (11), wherein the electronics unit (11) comprises filtering means for active sound transmission e.g. by amplification of chosen frequencies converted by said outer microphone (M1) and generating a corresponding acoustic signal through said sound generator (SG).

- Pick-up of the user's voice is performed by a microphone M2 with access to the closed space in the meatus 3. This signal is processed by means of analogue

or digital electronics in the electronics unit 11 to make it highly natural and intelligible, either for the user himself or his communication partners or both parts. This signal is of high quality and well suited for voice control and speech recognition.

- Online control and verification of the hearing protector performance is obtained by injecting an acoustic measurement signal, preferrably by the sound generator or loudspeaker SG in the meatus, and analysing the signal picked up by the microphone M2 that has access to the acoustic signal in the meatus 3.
- Measurement of noise exposure dose at the tympanum 4 and online calculation performed by electronic circuits, and warning of hearing damage risk either by audible or other warning signals, either to the wearer of the hearing protection or other relevant personnel.
- Equalisation of pressure between the two sides of the earplug system is obtained by using a very thin duct T3,T4 or a valve that equalises static pressure differences, while retaining strong low frequency sound attenuation. A safety valve V to take care of rapid decompression may be incorporated in the pressure equalisation system T3,T4.

Fig. 1 illustrates an embodiment according to the invention. The earplug comprises a main section 1 containing two microphones M1 and M2 and a sound generator SG. The main section is designed in a way that provides comfortable and secure placement in the concha (the bowl-shaped cavity at the entrance of the ear canal). This may be obtained by using individually moulded ear-pieces that are held in position by the outer ear or by using a flexible surrounding pressing against the structure of the outer ear. A sealing section 2 is attached to the main section. The sealing section may be an integral part of the earplug, or it may be interchangeable. The sound inlet of microphone M1 is connected to the outside of the earplug, picking up the external sounds. The microphone M2 is connected to the inner portion of the meatus 3 by means of an acoustic transmission

channel T1. The acoustic transmission channel may contain optional additional acoustic filtering elements. An outlet S_{SG} of sound generator SG is open into the inner portion of the meatus 3 by means of an acoustic transmission channel T2 between the sound generator SG and the inward facing portion of the sealing section 2. The acoustic transmission channel T2 may contain optional additional acoustic filtering elements.

When smaller microphones M2, and sound generators SG are available, it will be possible to mount the microphone M2 and the sound generator SG at the innermost part of the sealing section. Then there is no need for the transmission channels T1 and T2.

The two microphones and the sound generator are connected to an electronics unit 11, which may be connected to other equipment by a connection interface 13 that may transmit digital or analogue signals, or both, and optionally power.

Electronics and a power supply 12, e.g. a battery, may be included in main section 1 or in a separate section.

The microphones M1,M2 may in a preferred embodiment be standard miniature electret microphones like the ones used in hearing aids. Recently developed silicon microphones may also be used.

The sound generator SG may in a preferred embodiment be based on the electromagnetic or electrodynamic principle, like sound generators applied in hearing aids.

According to a preferred embodiment of the invention, a safety valve V is incorporated in the ventilation duct comprising the channels T3 and T4. The valve V is arranged to open if the static pressure in the inner part of the meatus 3 exceeds the outside pressure by a predetermined amount, allowing for pressure equalisation during rapid decompression. Such decompression may occur for military or civilian air personnel experiencing rapid loss of external air pressure. Such a decompression may also occur for parachuters, divers, and the like. Pressure equalisation for slowly varying pressure changes is obtained by using a narrow vent T4 which may bypass the valve V. A proper design

of this vent T4 allows for static pressure equalisation without sacrificing low-frequency noise attenuation.

The main section of the earplug may be made of standard polymer materials that are used for ordinary hearing aids. The sealing part may be made of a resilient, slowly re-expanding shape retaining polymer foam like PVC, PUR or other materials suitable for earplugs.

For some applications (less extreme noise levels) the earplug may be moulded in one piece 1,2 combining the main section 1 and the sealing section 2. The material for this design may be a typical material used for passive earplugs (Elacin, acryl).

It is also possible to make the earplug in one piece comprising the main section 1 and the sealing section 2, all made of a polymer foam mentioned above, but then the channels T1,T2,T3,T4 have to be made of a wall material preventing the channels T1,T2,T3,T4 to collapse when the sealing section 2 is inserted in the meatus 3.

All the features mentioned above may be obtained by an electric circuitry represented by the block diagram in Fig. 2.

The microphone M1 picks up the ambient sound. A signal from the microphone M1 is amplified in E1 and sampled and digitised in an analogue to digital converter E2 and fed to a processing unit E3 that may be a digital signal processor (DSP), a microprocessor (μP) or a combination of both. A signal 51 from microphone M2, which picks up the sound in the meatus 3 between the isolating section 2 and the tympanum 4, is amplified in the amplifier E4 and sampled and digitised in the analogue to digital converter E5 and fed to the processing unit E3.

A desired digital signal DS is generated in the processing unit E3. This signal DS is converted to analogue form in the digital to analogue converter E7 and fed to the analogue output amplifier E6 that drives the loudspeaker SG. The sound signal produced by the loudspeaker SG is fed to the tympanum 4 via the channel T2 into the meatus 3 as described above.

The processing unit E3 is connected to memory elements

RAM (Random access memory) E8, ROM (read only memory) E9, and EEPROM (electrically erasable programmable read only memory) E10. The memories E8,E9, and E10 are in a preferred embodiment of the invention used for storing computer programs, filter coefficients, analysis data and other relevant data.

The electronic circuitry 11 may be connected to other electrical units by a bi-directional digital interface E12. The communication with other electrical units may be performed via a cable or wireless through a digital radio link. The Bluetooth standard for digital short-range radio (Specification of the Bluetooth System, Version 1.0 B, 01 Dec 1999, Telefonaktiebolaget LM Ericsson) is one possible candidate for wireless communication for this digital interface E12.

In a preferred embodiment of the invention, signals that may be transmitted through this interface are:

- program code for the processing unit E3
- analysis data from the processing unit E3
- synchronisation data when two ear terminals 1,2 are used in a binaural mode
- digitised audio signals in both directions to and from an ear terminal 1,2.
- control signals for controlling the operation of the ear terminal.
- digital measurement signals for diagnosis of the ear terminal performance.

A manual control signal may be generated in E11 and fed to the processing unit E3. The control signal may be generated by operating buttons, switches, etc, and may be used to turn the unit on and off, to change operation mode, etc. In an alternative embodiment, a predetermined voice signal may constitute control signals to the processing unit E3.

The electric circuitry is powered by the power supply 12a that may be a primary or rechargeable battery arranged in the earplug or in a separate unit, or it may be powered via a connection to another equipment, e.g. a communication radio.

One embodiment of the invention concerns the use of the ear terminal as an "in-the-ear voice pick-up". The sound of a person's own voice as heard in the meatus is not identical to the sound of the same person's voice as heard by an external listener. The present embodiment of the invention remedies this problem. The microphone M2 illustrated in Fig. 3 picks up the sound in the inner portion of the meatus 3 sealed off by a sealing section 2 in an ear protecting communications device of the earplug type. The signal is amplified by the amplifier E4 illustrated in Fig. 2, A/D converted by the A/D converter E5, and processed in the digital signal processing (DSP) or microcomputer unit E3. The processing may be viewed as a signal dependent filtering taking into account the speech signal properties as well as computed estimates of the location of sound generation for the different speech sounds. Thereby the speech intelligibility and naturalness may be improved.

Figs. 1 and 3 show examples of embodiments of the invention, with the microphone M2 being integrated in a hearing protective communications earplug. The acoustic transmission channel T1 connects microphone M2 to the inner portion of the meatus 3. Microphone M2 picks up the sound field produced by the person's own voice. The signal may be amplified in amplifier E4, A/D converted in A/D converter E5 and processed in the digital signal processing (DSP) or microcomputer unit E3. A processed signal from E3 may be transmitted in digital form through a digital interface E12 to other electrical units. In an alternative embodiment, the processed signal from E3 may be D/A converted and transmitted in analogue form to other electrical units.

Fig. 4 illustrates one possible signal processing arrangement according to the invention. It illustrates an example of the type of signal dependent filtering which may be applied to the signal from microphone M2 in order to obtain a good reconstruction of the speech signal, making it highly intelligible, even in extremely noisy environments.

After amplification in E4 and A/D-conversion E5, the microphone M2 signal is analysed in the DSP/uP processing unit E3. The analysis represented by block 21 in Fig. 4 may comprise a short term estimate of the spectral power in the microphone signal, a short term auto-correlation estimate of the microphone signal, or a combination of both. Based on these estimates, a running classification with corresponding decision represented by block 22 may be made in the processing unit E3 for the selection of the most suitable conditioning filter for the signal from microphone M2. In the example shown in Fig. 4, the selection may made between e.g. three filters H1(f), H2(f) and H3(f) represented by blocks 23, 24 and 25, appropriate for vowel sounds, nasal sounds and fricative sounds respectively. The processed signal is present at output 26 of block 22. Other sound classifications using more sophisticated subdivisions between sound classifications and corresponding sound filters and analysis algorithms may be applied. The selection algorithm may comprise gradual transitions between the filter outputs in order to avoid audible artefacts. Filtering and selection is carried out in the processing unit E3 concurrently with the sound analysis and classification.

The basis for the filter characteristics and the corresponding analysis and classification in the processing unit E3 may be derived from an experiment of the form shown in Fig. 3. An ear plug with a microphone M2 with the same properties as the one used for the voice pickup is used to pick up the voice of a test subject from the meatus 3 illustrated in the upper part of Fig. 3. Concurrently, the voice is recorded by a high quality microphone M3 in front of the subject, at a nominal distance of 1 meter, under an-echoic conditions. Estimates of the power spectral densities may be computed for the two signals by the analyses represented by blocks 27 and 28 respectively, and the corresponding levels L1(f) and L2(f) are compared in

comparator 29. The output from the comparator is represented by the transfer function H(f). The analyses may be short time spectral estimates, e.g. 1/9 octave spectra in the frequency range 100 Hz to 14000 Hz. The test sequences which the subject utters may comprise speech sounds held constant for approximately 1 second. For voiced sounds, the subject person may make the pitch vary during the analysis period. The transfer functions of the filters described in connection with Fig. 4 may be based on diagrams of H(f), the spectral density levels of the free field microphone M3 subtracted from the corresponding levels of the in-the-ear microphone M2.

A simplest embodiment of the invention may reduce the system in Fig. 4 to one single time invariant filter. The analysis and selection processing may then be omitted. The transfer function of the single filter is still based on diagrams of the spectral density levels of the free field microphone subtracted from the corresponding levels of the in-the-ear microphone, described in connection with Fig. 3. The transfer function may be a combination of the results for the various speech sounds, weighted in accordance with their importance for the intelligibility and naturalness of the processed speech.

Another embodiment of the invention is best understood under the term "Natural Own Voice", indicating that a person wearing an ear terminal shall perceive his own voice as being natural while having the meatus blocked by an earplug.

The inner microphone M2 or the outer microphone M1, or a combination of both, picks up the sound signal representing the users voice signal. The signal is amplified, A/D converted, and analysed in the digital signal processor E3. Based on previously measured transfer functions from the user's speech to the microphone M2 (and/or M1), the microphone signal may be filtered to regain the naturalness of the user's speech. The signal is then

D/A-converted, amplified and reproduced at an internal loudspeaker SG. The internal loudspeaker SG may be arranged in a similar ear terminal 1,2 in the wearer's other ear to prevent local feedback in the earplug. In a more acoustically demanding arrangement the loudspeaker SG, arranged in the same meatus 3 as the inner pickup microphone M2 is situated, may be used, thus demanding feedback cancellation. The desired signal to the loudspeaker SG in the other ear may be transmitted via electric conductors outside of the wearer's head, or via radio signals.

Fig. 6 shows one preferred embodiment of the invention with the natural own voice feature being integrated in two active hearing protective communications earplugs. Each earplug may comprise a main section 1 containing two microphones, an outer microphone M1 and an inner microphone M2, and a sound generator SG. The right and left earplugs are generally symmetrical, otherwise identical for both ears. Section 2 is the acoustic sealing of the hearing protector. An acoustic transmission channel T1 connects microphone M2 to the inner portion of meatus 3. Microphone M2 picks up the sound from the meatus 3. When the user is speaking and the ear canal is sealed, this signal is mainly the user's own voice signal. This signal is filtered and reproduced at the loudspeaker SG at the other ear. An acoustic transmission channel T2 connects sound generator SG to the inner portion of meatus 3. A block diagram of the electronic system is shown in Fig. 2.

Fig. 4 shows an example of the type of signal dependent filtering which may be applied to the microphone signal in order to obtain a good reconstruction of the voice.

After amplification in E4 and A/D-conversion E5, the microphone M2 signal is analysed in the DSP/uP processing unit E3. The analysis represented by block 21 in Fig. 4 may comprise a short term estimate of the spectral power in the microphone signal, a short term auto-correlation estimate of the microphone signal, or a combination of both. Based on

these estimates, a running classification with corresponding decision represented by block 22 may be made in the processing unit E3 for the selection of the most suitable conditioning filter for the signal from microphone M2. In the example shown in Fig. 4, the selection may made between e.g. three filters H1(f), H2(f) and H3(f) represented by blocks 23, 24 and 25, appropriate for vowel sounds, nasal sounds and fricative sounds respectively. The processed signal is present at output 26 of block 22. Other sound classifications using more sophisticated subdivisions between sound classifications and corresponding sound filters and analysis algorithms may be applied. The selection algorithm may comprise gradual transitions between the filter outputs in order to avoid audible artefacts. Filtering and selection is carried out in the processing unit E3 concurrently with the sound analysis and classification.

The basis for the filter characteristics and the corresponding analysis and classification in the processing unit E3 may be derived from an experiment of the form shown in Fig. 5. An ear plug with a microphone M2 with generally the same properties as the one used for the voice pickup is used to pick up the voice of a test subject from the meatus 3 illustrated in the upper part of Fig. 5. Concurrently, the voice is recorded by a high quality microphone M4 close to the subject's ear, under an-echoic conditions. Estimates of the power spectral densities may be computed for the two signals by the analyses represented by blocks 37 and 38 respectively, and the corresponding levels L1(f) and L2(f) are compared in comparator 39. The output from the comparator is represented by the transfer function H(f). The analyses may be short time spectral estimates, e.g. 1/9 octave spectra in the frequency range 100 Hz to 14000 Hz. The test sequences which the subject utters may comprise speech sounds held constant for approximately 1 second. For voiced sounds, the subject may make the pitch vary during

the analysis period. The transfer functions of the filters described in connection with Fig. 4 may be based on diagrams of H(f), the spectral density levels of the free field microphone M4 subtracted from the corresponding levels of the in-the-ear microphone M2.

A simplest embodiment of the invention may reduce the system in Fig. 4 to one single time invariant filter. The analysis and selection processing may then be omitted. The transfer function of the single filter is still based on diagrams of the spectral density levels of the free field microphone subtracted from the corresponding levels of the in-the-ear microphone, described in connection with Fig. 5. The transfer function may be a combination of the results for the various speech sounds, weighted in accordance with their importance for the naturalness of the processed speech.

Another embodiment of the invention is called a "Personal Noise Exposure Dose Meter". Similar to the above embodiments, a microphone M2 picks up the sound in the meatus 3. One of the novel features is that this noise exposure is measured in the meatus, even while the ear is already noise protected. The signal from the microphone M2 is amplified, A/D converted, and analysed in a digital signal processing (DSP) or microcomputer unit E3 in the same way as described above. According to a preferred embodiment of the invention, the analysis covers both stationary or semistationary noise, and impulsive noise. The result of the analysis is compared to damage risk criteria and the user gets an audible or other form of warning signal when certain limits are about to be exceeded and actions have to be made. The warning signal may also be transmitted to other parties, e.g. industrial health care monitoring devices. The time record of the analysis may according to a preferred embodiment be stored in a memory, e.g. in the RAM E8 for later read-out and processing.

Fig. 1 shows a preferred embodiment of the invention with the personal noise exposure dose meter integrated in an active hearing protective communications earplug, comprising a main section 1 containing two microphones, an outer microphone M1 and an inner microphone M2, and a sound generator SG. Since this embodiment is part of a hearing-protecting earplug, a sealing section 2 is attached to the main section. An acoustic transmission channel T1 connects microphone M2 to the inner portion of the meatus 3. Microphone M2 therefore picks up the sound present in the meatus 3, just outside the eardrum (tympanum) 4. An acoustic transmission channel T2 connects sound generator SG to the inner portion of the meatus 3. The sound generator SG may provide audible information to the user, in form of warning signals or synthetic speech.

All the electronics as well as the battery are provided in the main section 1.

A block diagram of one possible implementation of this embodiment is shown in Fig. 2. The sound is picked up by the microphone M2, amplified, and AD-converted before it is fed to the processing unit E3 with DSP or uP (or both) as central processing units. The memory units E8 with RAM, E9 with ROM, and E10 with EEPROM may store programs, configuration data, and analysis results. Information to the user is generated in the central processing unit E3, DA-converted, amplified, and may be presented as audible information via the loudspeaker SG. The digital interface is used for programming, control, and readout of results.

The signal processing for the computation of noise exposure is shown in the flow diagram in Fig. 7. The signal from microphone M2 is amplified, converted to digital form and analysed by algorithms in processing unit E3. First, sample-by-sample equalization represented by block 41 is applied to compensate for irregularities in the microphone response, the transmission channel T1 and the missing ear canal response due to the blocking of the canal by the

earplug. The processed samples may according to the invention be evaluated in at least two ways. To obtain the stationary or semistationary noise dose, an A-weighting represented by block 42 is applied. Standards for this A-weighting exists: IEC 179, and the samples values are squared and accumulated in blocks 43 and 44 respectively. To obtain the peak value for assessing impulsive noise, a C-weighting represented by block 45 is applied according to internationally accepted standards, also IEC 179, and the peak value (regardless of sign) is saved in block 46. The noise dose and peak values are finally compared to predetermined limits in a decision algorithm represented by block 47 so that a warning may be given. The audible information to the user may be provided in form of warning signals or synthetic speech. The warning signal may also be transmitted to other parties, e.g. industrial health care monitoring devices. The time record of the two may also be stored in the memory of the processing unit E3 for later readout and further evaluation.

In addition to the use in passive ear protection devices this embodiment of the invention may be used as ear protection when the terminal is used as a headphone coupled to CD players for similar, monitoring the noise dose submitted from the headphones to the ear over time, or in peaks.

Another embodiment of the invention, called "Online verification/control of hearing protector performance", utilises the fact that a sound field locally generated in the cavity near the ear drum is influenced by leakage in the hearing protector. A small electro-acoustic transducer (sound source) SG and a microphone M2 are arranged in a sealing section 2 arranged for attenuating sounds entering the meatus cavity 3. A digital signal processing (DSP) or microcomputer unit E3 in the main section 1 or in the sealing section 2 is used to generate a predetermined signal

which is D/A converted by the D/A converter E7, amplified by amplifier E6 and applied to the sound source SG, which generates a sound field in the closed part of the meatus 3. The microphone M2 picks up the sound in the meatus cavity 3. This signal is amplified by amplifier E4, A/D converted by A/D converter E5, and analysed in the digital signal processor or microprocessor E3. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions. The user may get audible or other messaged confirmation if the leakage is acceptably low, or a warning signal if leakage is unacceptably high. In the same manner, a signal may be transmitted to other instances, e.g. an external industrial health monitoring unit, with information about the leakage. One example may be that an ear terminal according to the invention is used for checking for leakage in the hearing protection while the wearer is at a gate controlling admittance to a noise exposed area. If leakage occurs, a signal may be transmitted from the ear terminal to a corresponding signal receiver at the gate, having means to block the gate for entrance until the leakage condition is remedied and verified.

Fig. 1 illustrates an embodiment of the invention where the verification device is integrated in a hearing protective earplug. This embodiment comprises an outer section 1 containing a microphone M2 and a sound generator SG. An inner sealing section 2 is attached to the outer section, but may be made in one integrated outer section/sealing section 1,2. An acoustic transmission channel T2 connects sound generator SG to the inner portion of the meatus 3. The sound generator SG produces a predetermined acoustic signal, which generates a sound field in the meatus 3. An acoustic transmission channel T1 connects microphone M2 to the inner portion of the meatus 3. Microphone M2 picks up the sound field being set up by the sound generator SG. The signal generation and analysis is

carried out in a digital signal processing (DSP) or microcomputer unit E3 with appropriate amplifiers and converters as described in the previous paragraphs. All the electronics 11 as well as the power supply 12 are provided in the outer section 1.

Fig. 8 illustrates a signal processing arrangement according to a preferred embodiment of the invention. This embodiment utilises a signal which produces reliable characterisation of the sound field in the cavity, preferably while not being annoying to the user. The signal may comprise one or more sinusoidal components presented simultaneously, or in sequence. Alternatively, a pseudorandom sequence may be employed. In both cases, preferably both the in-phase and the out-of-phase portions of the sound field are analysed and used in the verification algorithm.

An example of the signal processing is shown in the flow diagram in Fig. 8. In the example, two pure tones of different frequencies f_1 and f_2 are generated by algorithms in the processing unit E3. The generators are represented by blocks 81 and 82 respectively. The generators generate both the in-phase (sin) and out-of-phase (cos) components. The in-phase components are added together in block 83, converted to analogue form, amplified and applied to the sound generator SG. The resulting sound field is picked up by the microphone M2, amplified, converted to digital form and analysed by algorithms in the processing unit E3 for a series of detectors represented by blocks 84, 85, 86 and 87. The in-phase and out-of-phase components of the microphone M2 signal are analysed for each of the two frequencies. The detector algorithm performs a sample-by-sample multiplication of the two input signals and smoothes the result with a low-pass filter. The four detector outputs are applied to a decision algorithm represented by block 88 where they are compared to stored values. The decision result may be a digital "go" / "no go" real time signal

indicating acceptable noise protection attenuation or unacceptable protection conditions. The result of the analysis is compared to stored results from previous measurements of the same type in a situation with good sealing conditions.

The stored values for the decision algorithm may according to a preferred embodiment be based on previous laboratory experiments, but values for the decision algorithm may also be determined, e.g. making an average and setting a lower acceptance limit for a general-purpose embodiment of the invention.

The number and values of frequencies and the smoothing characteristics of the detectors are chosen as a compromise between audibility and response time. If a continuously running verification should be necessary, low frequencies, e.g. in the range of 10-20 Hz, of sufficiently low levels may be utilised in order to avoid annoyance. The pure tones may then be partially or fully aurally masked by the residual noise transmitted by the hearing protector.

The acoustic phenomenon on which the embodiment of the invention is based is illustrated by the electric analogy diagram in Fig. 9. In the diagram, the sound generator SG is modelled by its acoustic Thevenin equivalent represented by blocks 91 and 92. The sound pressure p1 is generated by the Thevenin generator 91, resulting in a volume velocity through the Thevenin impedance Z1(f) 92. The microphone M2 is modelled by its acoustic impedance Z3(f) represented by block 93. The sound pressure p2 at the microphone entrance is converted to an electric signal by the microphone. For the purpose of the present illustration, all acouostic elements exposed to the sound pressure generated by the sound generator SG, except the microphone, are lumped together in the acoustic impedance Z2(f) represented by block 95. A leakage in the hearing protector may be modelled by a change in the variable acoustic impedance Z2(f). The change will usually affect both the frequency dependent

modulus and the frequency dependent phase of Z2(f). This change leads to a change in the relationship between the sound pressures p2 and p1, which is analysed as described in connection with Fig. 8.

C l a i m s

1. Ear terminal comprising a sealing section (2) for arrangement in the meatus (3) of a human, comprising:

an inner microphone (M2) having a sound inlet (S2) for being directed directed into the meatus,

an electronics unit (11) coupled the inner microphone (M2) and also being coupled to a power supply (12) as well as

an outer microphone (M1) for converting acoustic signals in the environment into electrical signals.

2. Ear terminal according to claim 1, comprising a pressure alignment channel (T3) for slow air throughput to and from the meatus (3) through the sealing section (2).

3. Ear terminal according to claim 1, wherein the sound inlet (S2) is constituted by a canal (T2) between the microphone (M2) and the inward facing portion of the sealing section (2) and the meatus.

4. Ear terminal according to claim 2, wherein the pressure alignment channel (T3) includes a pressure release valve (V) arranged for opening if the pressure difference between the meatus and the environment of the user exceeds a predetermined limit.

5. Ear terminal according to claim 4, comprising a bypass channel (T4) in the pressure alignment channel (T3).

6. Ear terminal according to claim 1, comprising a connection interface (13,E12), e.g. a radio receiver or electric coupling, coupled to the electronics unit (11), said electronics unit being provided with conversion means (E7) for converting signals received from said interface and

being coupled to the sound generator (SG) for transmitting acoustic information to the user.

7. Ear terminal according to claim 1, comprising a connection interface (13,E12), e.g. a radio transmitter or electric coupling, coupled to the electronics unit (11), said electronics unit being provided with conversion means (E5) for converting signals received from the inner microphone (M2) for transmitting electric or electromagnetic information from the user.

8. Ear terminal according to claim 1, comprising a sound generator (SG) coupled to said electronics unit (11), wherein the electronics unit comprises analyzing means for active noise cancelling, e.g. by feedback of acoustic signals converted by at least one of said microphones (M1,M2) generated through said sound generator (SG).

9. Ear terminal according to claim 1, comprising a sound generator (SG) arranged for being directed toward the meatus and being coupled to said electronics unit (11), wherein the electronics unit (11) comprises filtering means for active sound transmission e.g. by amplification of chosen frequencies converted by said outer microphone (M1) and generated a corresponding acoustic signal through said sound generator (SG).

10. Ear terminal according to claim 1, comprising a sealing section (2) arranged for use in the ear meatus (3) of a human, in which the electronic unit (11) including filtering means coupled to said inner microphone for filtering the signal from said inner microphone (M2), said filtering means being programmable to transform the signals based on the sounds received in the ear by said inner microphone (M2) into sounds having essentially the characteristics of spoken sounds of the wearer of the ear terminal.

11. Ear terminal according to claim 10, comprising a connection interface (13,E12) for transmitting the filtered signal from the ear terminal;

12. Ear terminal according to claim 1, for protection of a users hearing comprising a sealing section (2) for acoustically sealing the meatus (3) of a human, comprising

a sound generator (SG) with a sound outlet (S_{SG}) for being directed toward the user meatus (3);

the electronics unit (11,E3) comprising a sound analyser coupled to said inner microphone (M2), for analyzing sound characterisics of the resulting sound field in the meatus (3), producing analyzed sound characteristics;

storing means in the elecronics unit (11,E8,E9,E10) for storing measured predetermined sound characteristics of a properly functioning ear protecting device;

a comparing means in the electronics unit (11,E3) for comparing the inner microphone (M2) analyzed sound characteristics with the stored measured predetermined sound characteristics;

indicating means coupled to said comparing means (11,E3) for being activated if said analyzed sound characterisics differ significantly from said predetermined sound characteristics.

13. Ear terminal according to claim 1 comprising a sealing section (2) arranged for use in the ear meatus (3) of a human, and

an electronic unit (11) including filtering means coupled to said inner microphone for filtering the signal from said inner microphone (M2), said filtering means being programmable to transform the signals based on the sounds received in the ear by said inner microphone (M2) into sounds when combined with the users own voice, providing a voice sounding natural to the user.

14. Ear terminal according to claim 13, comprising a sound generator (SG) having a sound outlet arranged for being directed into the meatus, said electronic unit also comprising feedback suppresion means for suppressing the feedback between the sound generator (SG) and the inner microphone (M2).

15. Ear terminal system comprising two ear terminals, at least the first of which being an ear terminal according to claim 13, said first ear terminal comprising a connection interface (13,E12) for transmitting the filtered signal from the ear terminal; into said second ear terminal arranged for being positioned in the opposite ear of the user, said second ear terminal comprising a corresponding connection interface (13,E12) and a sound generator (SG) for emitting the sound into said second ear.

16. Ear terminal according to claim 1, wherein the inner microphone (M2) has a sound inlet (S2) for being directed toward the meatus (3), arranged for converting a picked-up sound signal (52) to an output signal (51); the ear protecting device also comprising a an electronics unit (11) including sound analyzing means arranged for analyzing the output signal (51) and comparing the signal with predetermined signals corresponding to acceptable noise limits and for activating an indicator (72) when detecting a passing of these limits.

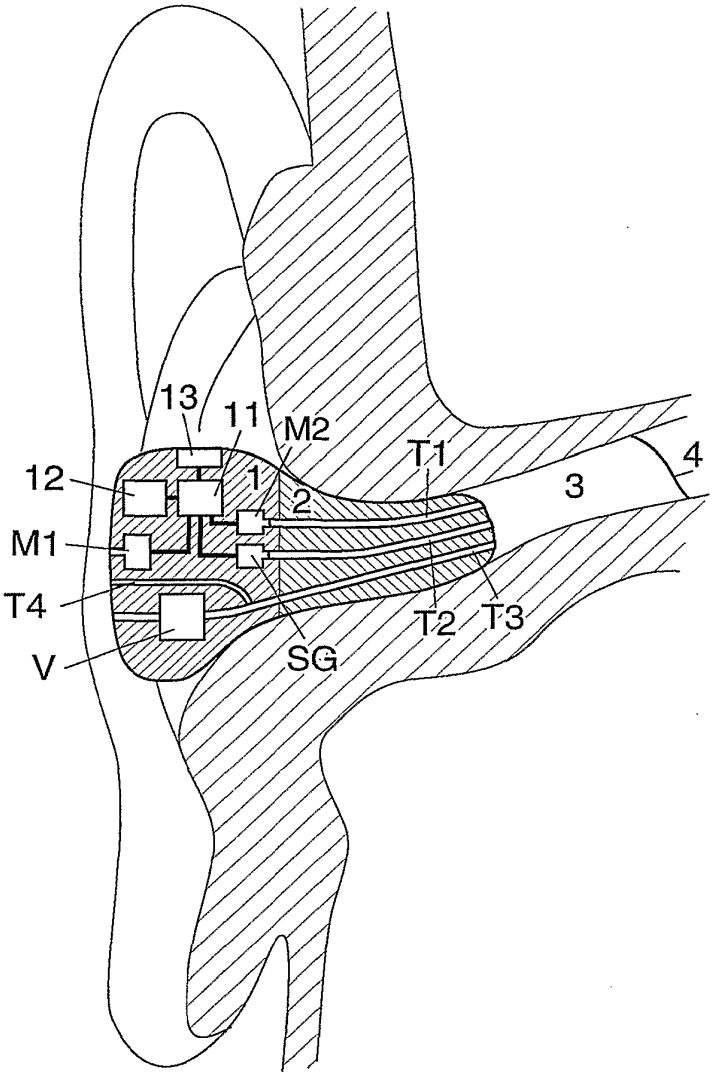


Fig. 1

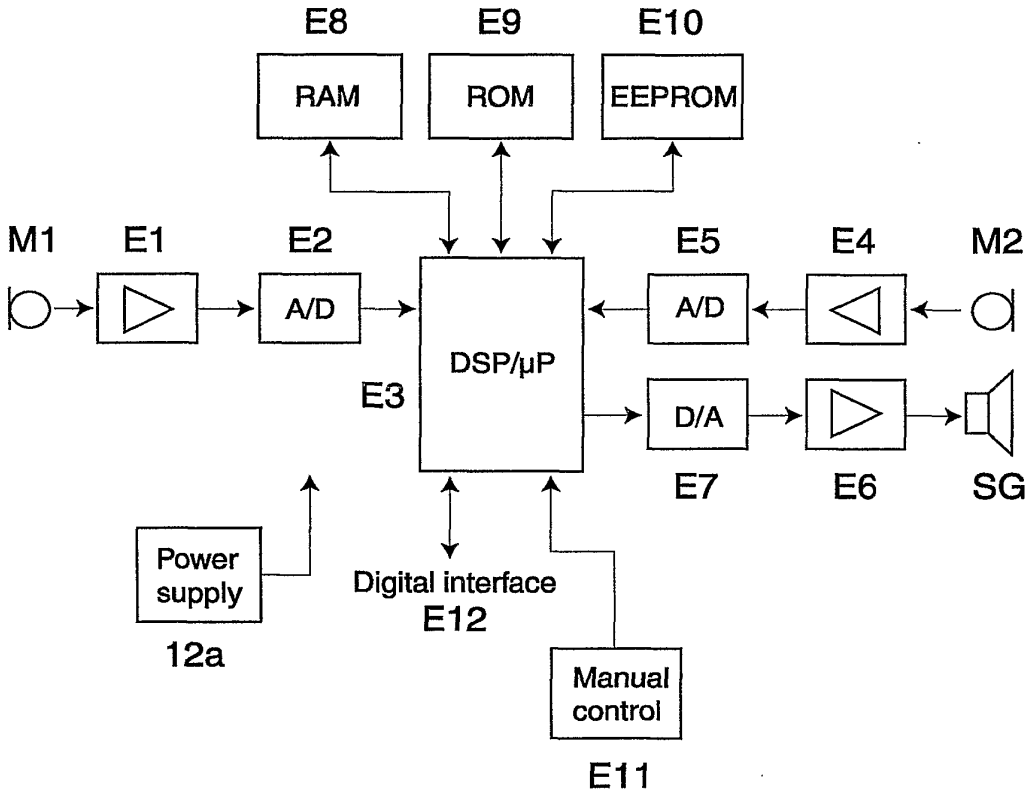

E8
E9
E10
RAM
ROM
EEPROM
M1
E1
E2
E5
E4
M2
A/D
A/D
DSP/µP
E3
D/A
E7
E6
SG
Power supply
12a
Digital interface
E12
Manual control
E11

Fig. 2

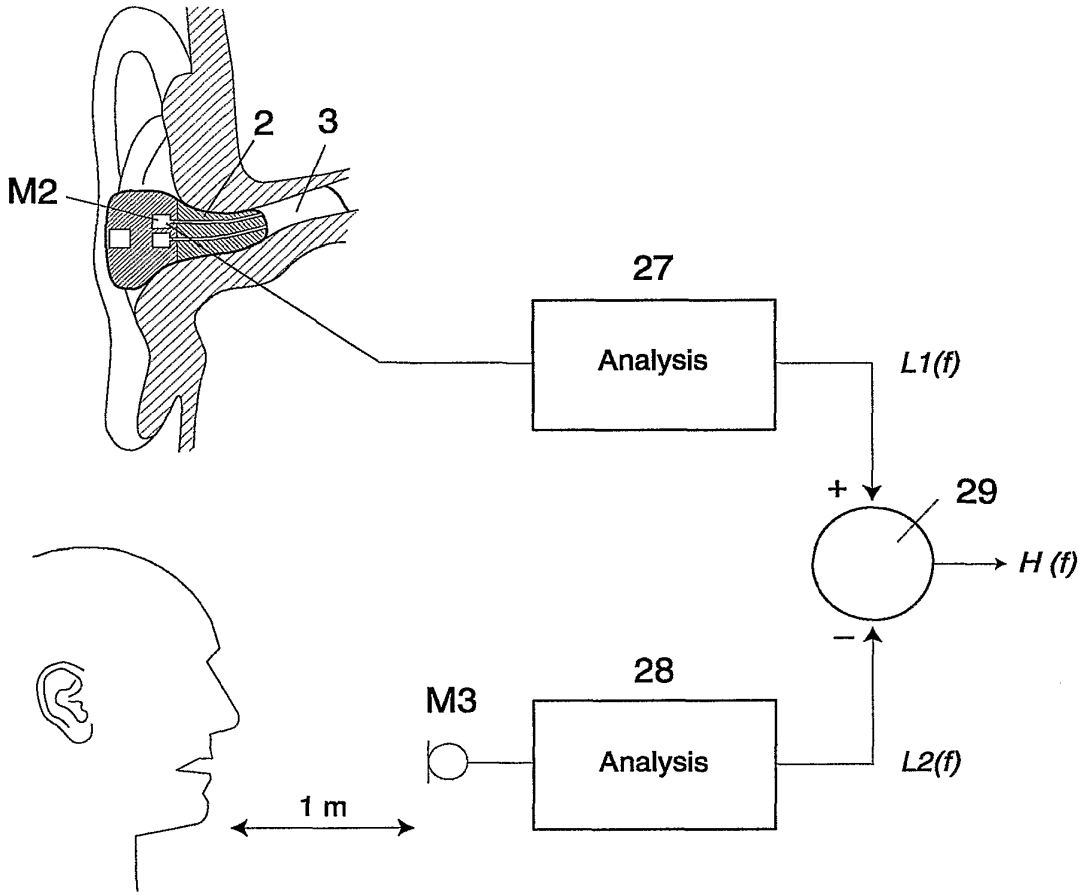


Fig. 3

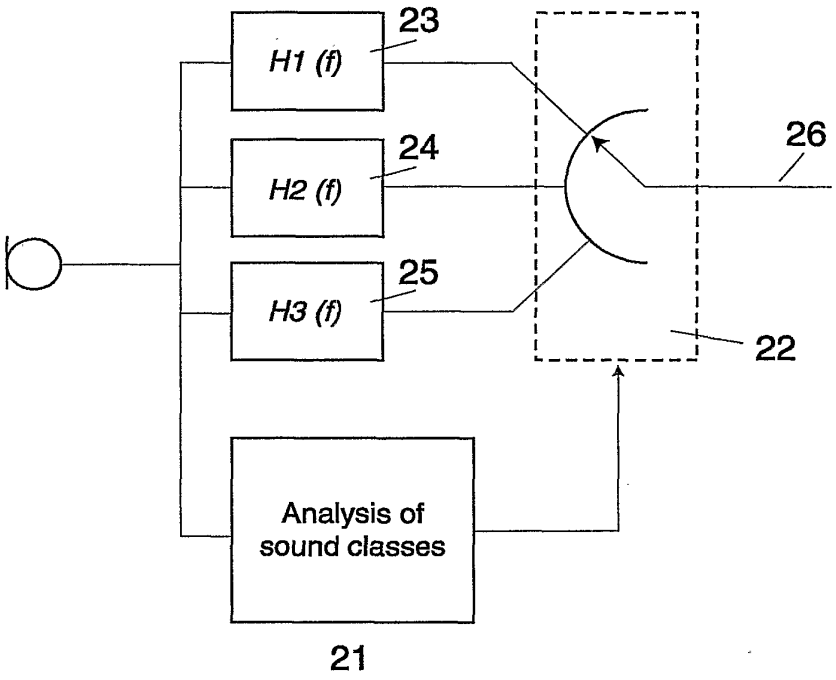


Fig. 4

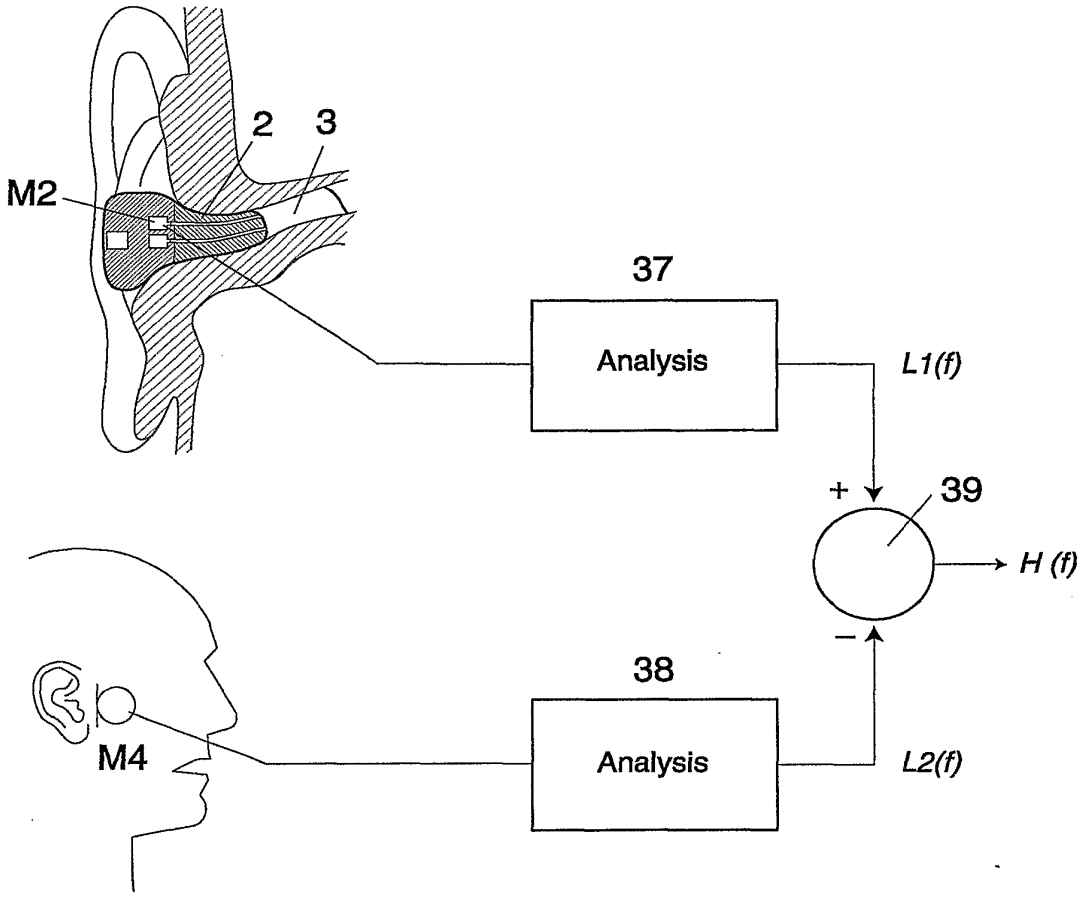


Fig. 5

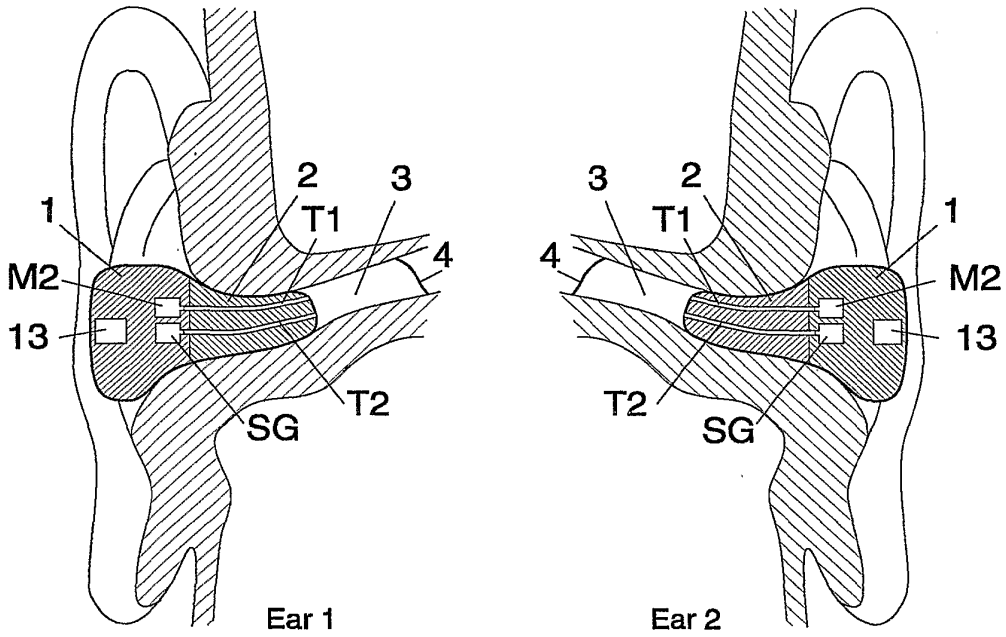


Fig. 6

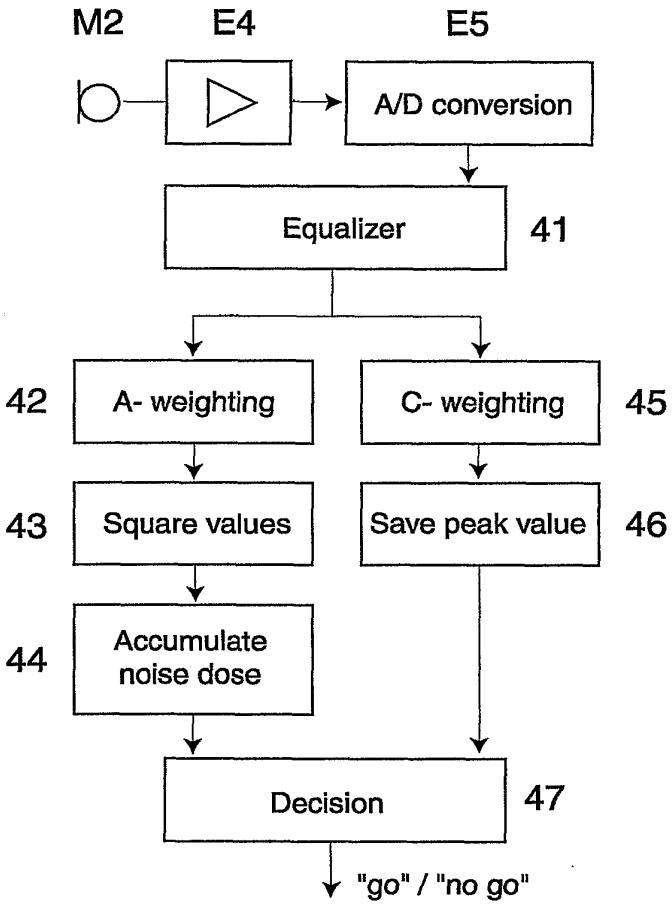

M2
E4
E5
A/D conversion
Equalizer
41
42
A- weighting
C- weighting
45
43
Square values
Save peak value
46
44
Accumulate noise dose
Decision
47
"go" / "no go"

Fig. 7

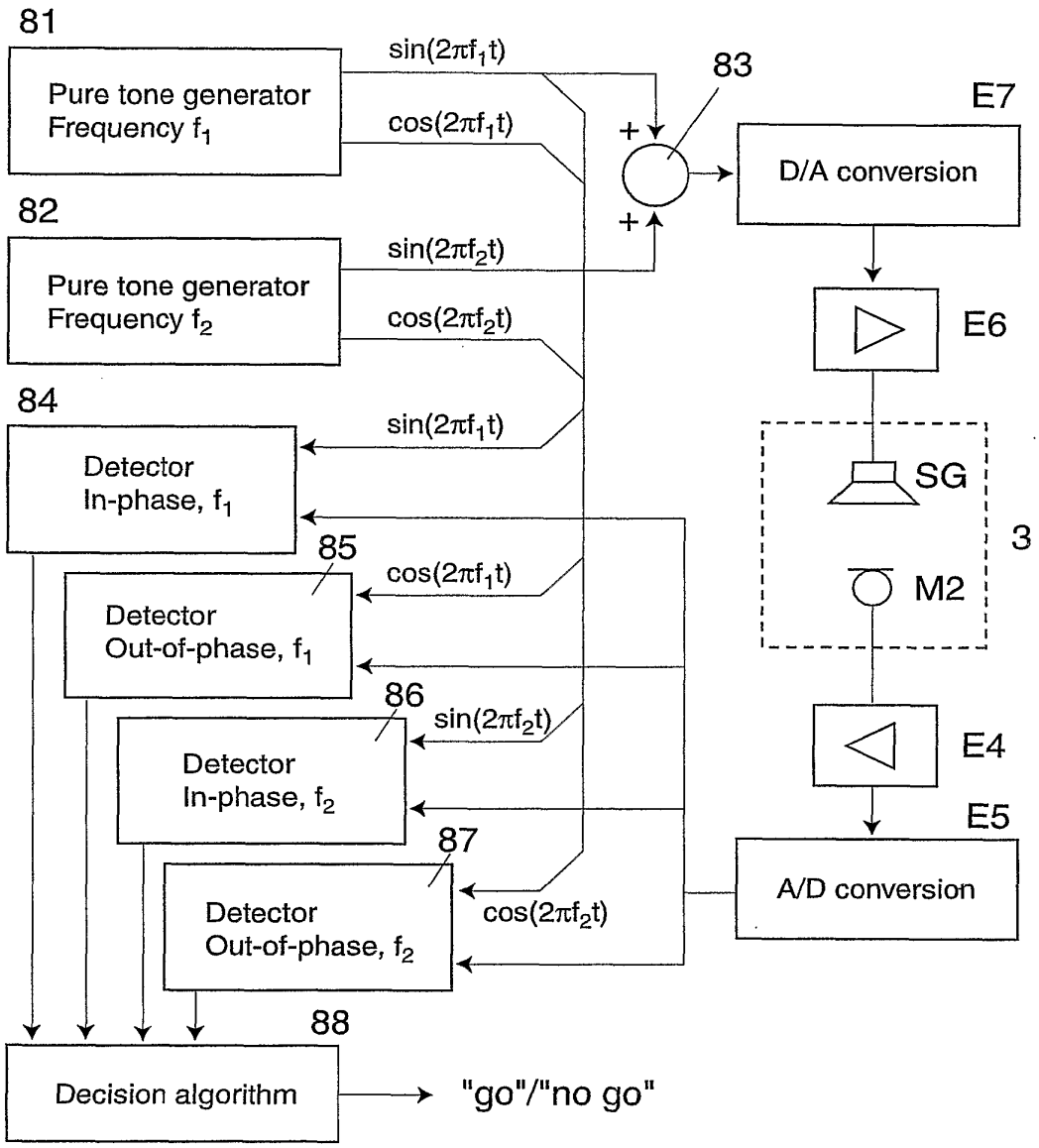

81
Pure tone generator
Frequency f_1
$\sin(2\pi f_1 t)$
$\cos(2\pi f_1 t)$
83
+
+
E7
D/A conversion
82
Pure tone generator
Frequency f_2
$\sin(2\pi f_2 t)$
$\cos(2\pi f_2 t)$
E6
84
Detector
In-phase, f_1
$\sin(2\pi f_1 t)$
SG
3
M2
85
Detector
Out-of-phase, f_1
$\cos(2\pi f_1 t)$
86
Detector
In-phase, f_2
$\sin(2\pi f_2 t)$
E4
E5
87
Detector
Out-of-phase, f_2
$\cos(2\pi f_2 t)$
A/D conversion
88
Decision algorithm
"go"/"no go"

Fig. 8

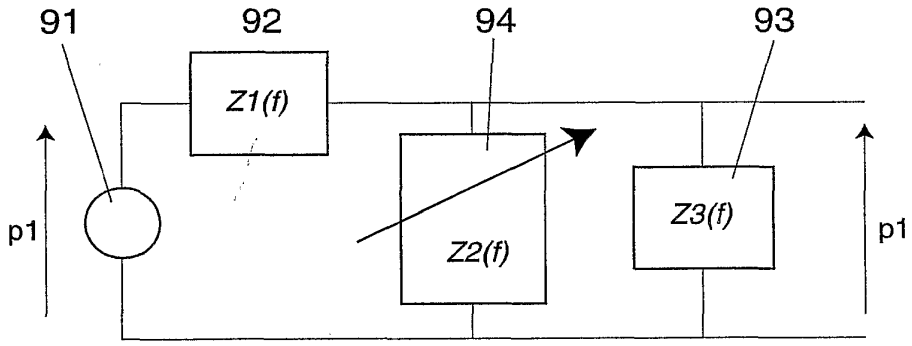


Fig. 9

INTERNATIONAL SEARCH REPORT

International application No.
PCT/NO 01/00357

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: A61F 11/08, H04R 25/02 // A61F 11/06, H04R 25/00
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)

IPC7: A61F, H04R

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

SE,DK,FI,NO classes as above

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

EPO-INTERNAL, WPI DATA

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5577511 A (KILLION, MEAD C.), 26 November 1996 (26.11.96), whole document	1
Y	--	2-11,13-16
Y	US 5757930 A (SEIDEMANN, MICHAEL F. ET AL.), 26 May 1998 (26.05.98), abstract	1-11,13-16
	--	
Y	EP 0684750 A2 (ERMES S.R.I.), 29 November 1995 (29.11.95), whole document	1-11,13-16
	--	
A	US 5327506 A (STITES, III, GEORGE M.), 5 July 1994 (05.07.94)	1-16
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[X] Further documents are listed in the continuation of Box C. [X] See patent family annex.

* Special categories of cited documents:
"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier application or patent but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
19 December 2001	21 -12- 2001
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer Henrik Andersson/LR Telephone No. +46 8 782 25 00

Form PCT/ISA/210 (second sheet) (July 1998)

INTERNATIONAL SEARCH REPORT

International application No.
PCT/NO 01/00357

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5317273 A (HANSON, WILLIAM J. ET AL.), 31 May 1994 (31.05.94) --	1-16
A	US 4985925 A (LANGBERG, EDWINET AL.), 15 January 1991 (15.01.91) --	1-16
A	US 5426719 A (FRANKS, JOHN R. ET AL.), 20 June 1995 (20.06.95) -- --------	1-16

Form PCT/ISA/210 (continuation of second sheet) (July 1998)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No.
PCT/NO 01/00357

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
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US 5757930 A	26/05/98	US 5970795 A	26/10/99
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		US 5535282 A	09/07/96
US 5327506 A	05/07/94	AU 4368493 A	21/11/94
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		US 5208867 A	04/05/93
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US 4985925 A	15/01/91	AT 147961 T	15/02/97
		DE 68927707 D,T	07/05/97
		EP 0423172 A,B	24/04/91
		JP 2989843 B	13/12/99
		JP 3505639 T	05/12/91
		WO 8912432 A	28/12/89
US 5426719 A	20/06/95	AU 5096793 A	29/03/94
		WO 9405231 A	17/03/94

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Office européen des brevets

(11) EP 1 519 625 A2

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(54) External ear canal voice detection

(57) Ear-level full duplex audio communication systems each include one or two ear attachment devices, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, that wirelessly communicates to a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. When used as a hearing aid, such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. When being used as an ear piece wirelessly extended from the remote device, such system allows an individual with normal hearing to privately communicate with or through the remote device without the need of holding the device or wearing any device wired to the remote device. Each ear attachment device includes a voice operated exchange (VOX), housed within the device, to preserve energy and hence, maximize the period between battery replacement or recharges. The VOX also gates various sounds detected by the system to control possible echoes and ringing.

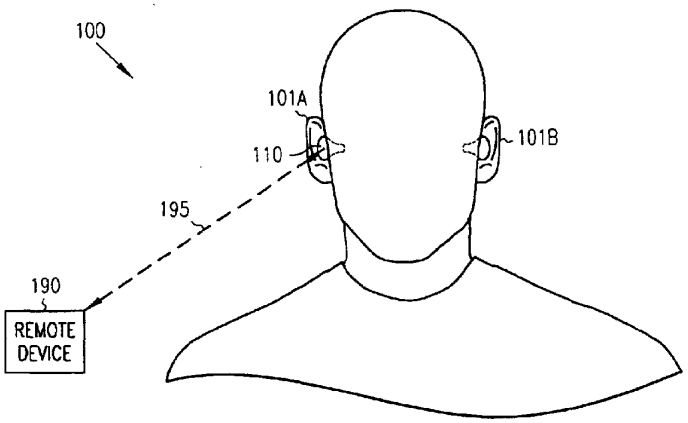


FIG. 1

EP 1 519 625 A2

Printed by Jouve, 75001 PARIS (FR)

Description

[0001] This document generally relates to ear-level communication systems and particularly, but not by way of limitation, to such systems communicating with remote devices via full duplex audio.

[0002] Ear-level communication systems are used for hearing impaired individuals to use a communication device such as a cellular phone or other devices transmitting audio signals. They also allow private, hand-free use of such a communication device by individuals with normal hearing.

[0003] An ear-level communication system generally includes tethered or wireless headsets with a microphone boom or a microphone within a cord of the headset to detect voice of the person wearing the headsets. An alternative way to detect voice is via bone connection. Examples include a "clam" shell device that couples to the helix and a custom carmold with a vibration detector on the outer shell. Voice is picked up by detecting bone vibration caused by, and representative of, the voice of the person wearing the device.

[0004] Bi-directional communication has been achieved by a simplex system including a device in the ear canal that both transmits sound to and picks up sound within the ear canal. The device is tethered to a push-to-talk (PIT) system such as a walkie talkie. It is not full duplex and therefore requires interaction by the user in order to switch from listen mode to talk mode.

[0005] Voice operated exchange (VOX) is used to conserve energy when voice is not being detected or transmitted. For example, a headset system includes two ear pieces - for full duplex communication with one earpiece for transmitting and the other for receiving voice - and uses ear canal voice detection to trigger a VOX circuit in a control unit wired to the headset. The system requires a headset with two headphones and wires connecting the headphone to a control unit such as a device clipped on belt. In another example, a full duplex system includes a single headphone to transmit and receive voice, again with a VOX circuit in a control unit wired to the headphone.

[0006] Such ear-level communication systems are likely to be worn by a person for long periods of time. Users with hearing loss may use such a system on a regular basis, both as a conventional hearing aid allowing communication directly with a person and as a hearing aid allowing communicating through another device. Users with normal hearing may wear such a system to be communicatively connected to another person or facility without the need to hold a device at any time. For appearance or secrecy reasons, minimal visibility is generally desirable. Wired connection between a headset or earpiece and a controller makes the system easily visible and is cumbersome to users who need to communicate white being physically active.

[0007] Thus, there is a need for an ear-level communication system that is not easily visible. A need related to miniaturization is the longevity of the system between battery replacements or recharges.

[0008] Ear-level full duplex audio communication systems each include one or two ear attachment devices, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, that wirelessly communicates to a remote device. Such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. Such a system also functions as an ear piece wirelessly extended from the remote device that allows an individual to privately communicate with or through the remote device without the need of holding the device.

[0009] In one embodiment of the invention, a system includes an earmold configured for use as an ITE device. The earmold houses a microphone for use in an ear canal, a processor, and a wireless transmitter. The microphone receives an occluded sound from about the ear canal. The processor processes the occluded sound. The wireless transmitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0010] In one embodiment, a system includes an ITE module and a BTE module attached to the ITE module. The ITE module includes a microphone for use in an ear canal. The microphone receives an occluded sound from about the ear canal. The BTE module includes a processor and a wireless transmitter. The processor processes the occluded sound. The wireless transmitter receives the processed occluded sound from the processor and transmits a wireless signal representing the occluded sound.

[0011] In one embodiment, a system includes a first ear-level device and a second ear-level device for use in two opposite ears. The first ear-level device includes a microphone, a first processor, and a wireless transmitter. The microphone is for use in one ear canal to receive an occluded sound from about that ear canal. The first processor converts the occluded sound to an electrical signal. The wireless transmitter receives the electrical signal and transmits an outgoing wireless signal representing the occluded sound. The second ear-level device includes a wireless receiver, a second processor, and a speaker. The wireless receiver receives an incoming wireless signal representing a remote sound. The second processor converts the incoming wireless signal to the remote sound. The speaker is for use in the opposite ear canal to transmit the remote sound to that ear canal.

[0012] In one embodiment, an occluded sound is detected using a microphone placed in an ear canal. An incoming radio signal representing a remote sound from a remote device is also detected. A voice operated exchange (VOX) housed within an ear-level communication device is started when at least one of the occluded sound and the incoming radio signal is detected if the VOX is not already on. The ear-level communication device includes one of an ITE device and a BTE device.

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When the occluded sound is detected while the incoming radio signal is not detected, the occluded sound is gated on, and remote sound is gated off. When the incoming radio signal is detected while the occluded sound is not detected, the remote sound is gated on, and the occluded sound is gated off.

[0013] Preferred embodiments of the present invention will now be described by way of example only and with reference to the accompanying drawings, in which:

PIG. 1 is an illustration of an embodiment of an ear-level communication system and portions of an environment in which it is used. This ear-level communication system uses a single ear-level device.

FIG. 2 is an illustration of an embodiment of another ear-level communication system and portions of an environment in which it is used. This ear-level communication system uses a pair of ear-level devices.

FIG. 3A is an illustration of one exemplary exterior configuration of an in-the-ear (ITE) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG, 2.

FIG. 3B is an illustration of one exemplary exterior configuration of an in-the-canal (TTC) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 3C is an illustration of one exemplary exterior configuration of a completely-in-the-canal (CIC) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device of FIG. 1.

FIG. 4B is a block diagram showing one embodiment of portions of a voice operated exchange (VOX) circuit included in the single car-level device of FIG. 1.

FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the pair of ear-level devices of FIG. 1B.

FIG. 6 is an illustration of one exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device used as the single ear-level device of FIG. 1 or one of the pair of ear-level devices of FIG. 2.

FIG. 7 is a block diagram showing one embodiment of the circuit of the ear-level device shown in FIG. 6 used as the single ear-level device of FIG. 1.

FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as the pair of ear-level devices of FIG. 2.

FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication between the single ear-level device or the pair of ear-level devices and a remote device.

FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication between the single ear-level device and a remote device.

[0014] The following embodiments are described in sufficient detail to enable those skilled in the art to practice the invention, and it is to be understood that the embodiments may be combined, or that other embodiments may be utilized and that structural, logical and electrical changes may be made without departing from the scope of the present invention, as defined by the appended claims.

[0015] It should be noted that references to "an", "one", or "various" embodiments in this disclosure are not necessarily to the same embodiment, and such references contemplate more than one embodiment.

[0016] This document discusses, among other things, ear-level full duplex audio communication systems for communicating with remote devices. These systems each include one or two ear-level devices each worn in or about an ear, such as in-the-ear (ITE) or behind-the-ear (BTE) devices, wirelessly coupled to a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. When used as a hearing aid, such a system allows a hearing impaired individual to communicate with or through the remote device, such as to talk to another person through a cellular phone. When being used as an ear piece wirelessly extended from the remote device, such system allows an individual with normal hearing to privately communicate with or through the remote device without the need of holding the device or wearing any device wired to the remote device.

[0017] In this document, "voice" includes to the sound of speech made by a person, and sound generally includes an audible signal that includes voice and other signals that can be heard and/or detected by an audio detector. Unless particularly indicated, for example, when voice is distinguished from sound by speech recognition, "voice" and "sound" are used interchangeably throughout thus document because they are not practically separated. For example, detection of a sound refers to the same action as detection of a voice because, while the purpose is to detect the voice, other sounds are picked up as well.

[0018] FIG. 1 is an illustration of an embodiment of an ear-level communication system 100 and portions of an environment in which it is used. System 100 includes an ear-level device 110 attached to an ear 101A. Bar-level device 110 communicates with a remote device 190 through a wireless telemetry link 195. The other ear 101B is free of device attachment. Ear-level device 110 is a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of ear-level device 110 is voice activated. That is, ear-level device 110 is activated only when it detects a need to transmit or receive a

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sound.

[0019] Ear-level device 110 is a full duplex audio device that allows two-way simultaneous conversation between ear 101A and remote device 190. In one embodiment, when being worn by a person having ear 101A, ear-level device 110 picks up sound from the ear canal of ear 101A and delivers sound to the same ear canal. It detects the occluded sound from the ear canal when the person speaks and transmits to the same ear canal a sound received from remote device 190 and/or a sound picked up from the environment surrounding the person. In one embodiment, as illustrated in FIG. 1, ear-level device 110 is configured for right ear attachment. In an alternative embodiment, ear-level device 110 is configured for left ear attachment.

[0020] FIG. 2 is an illustration of an embodiment of an ear-level communication system 200 and portions of an environment in which it is used. System 200 includes a pair of ear-level devices 210A, attached to an ear 101 A, and 210B, attached to ear 101B. Ear-level devices 210A and 210B communicate with a remote device 290 through a wireless telemetry link 295. Ear-level devices 210A and 210B are each a self-contained device including its own power source such as a replaceable or rechargeable battery. To maximize battery life or period between recharges, at least a portion of the circuit of each of ear-level devices 210A and 210B is voice activated. That is, ear-level devices 210A and 210B are each activated only when a need to transmit or receive a sound is detected by the device itself.

[0021] In one embodiment, ear-level device 210A is an audio transmitter that picks up sound from the ear canal of ear 101A, and ear-level device 210B is an audio receiver that delivers sound to the same ear canal. Ear-level device 210A is activated when it detects a sound from the ear canal. Ear-level device 210B is activated when it detects a signal from remote device 290. When both are being worn by a person, ear-level devices 210A and 210B supports full duplex audio allowing two-way simultaneous conversation between the person and remote device 290. In one embodiment, as illustrated in FIG. 2, ear-level device 210A is configured for right ear attachment, and ear-level device 210B is configured for left ear attachment. In an alternative embodiment, ear-level device 210A is configured for left ear attachment, and ear-level device 210B is configured for right ear attachment.

[0022] FIGS. 3A, 3B, and 3C are illustrations of exemplary exterior configuration of ear-level device 110 or one of ear-level devices 210A and 210B. For the purpose of this document, only a device attached to a right ear is shown for each exemplary exterior configuration. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a device configured for right ear attachment or a device configured for left ear attachment. Generally, a device configured for right car attachment has an exterior configuration that is substantially symmetrical to that of a device of the same type configured for left ear attachment. In system 100, a single ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of car-level device 210A and 210B is a device configured for right ear attachment, while the other is a device configured for left ear attachment.

[0023] An occluded sound 305, an ambient sound 306, and a remote sound 307 are illustrated in each of FIGS. 3A, 3B, and 3C. Remote sound 307 is represented by a wireless radio signal transmitted to ear-level device 110. In this document, "occluded sound" includes occluded sound 305, "ambient sound" includes ambient sound 306, and "remote sound" includes remote sound 307.

[0024] FIG. 3A is an illustration of an exemplary exterior configuration of an in-the-ear (ITE) device 310A used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3A, ITE device 310A is attached to an ear 101 with a pinna 302 and an ear canal 303. ITE device 310A includes a portion inserted into ear canal 303 and another portion fit into the cavity formed by pinna 302.

[0025] FIG. 3B is an illustration of an exemplary exterior configuration of an in-the-canal (ITC) device 310B used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3B, ITC device 310B is attached to ear 101 with pinna 302 and ear canal 303. ITC device 310B is a form of ITE device that also includes a portion inserted in ear canal 303 and another portion fit into the cavity formed by pinna 302, but the latter portion is smaller than that of ITE device 310A. Its overall size is therefore generally smaller than that of ITE deviee 310A.

[0026] FIG. 3C is an illustration of an exemplary exterior configuration of a completely-in-the-canal (CIC) device 310C used as ear-level device 110 or one of ear-level devices 210A and 210B. As shown in FIG. 3C, CIC device 310C is attached to ear 101 with pinna 302 and ear canal 303. CIC device 310C is a form of ITC device but includes only a portion that is completed inserted into ear canal 303. Its overall size is therefore generally smaller than that of ITC device 310B.

[0027] In one embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a custom earmold. In another embodiment, each of ITE device 310A, ITC device 310B, and CIC device 310C includes a modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0028] The choice of a device type, such as from the types shown in FIGS. 3A, 3B, and 3C, depends on the need of individual users. For example, ITE device 310A has a relatively large size to accommodate a relatively large amplifier to serve the need of individuals suffering severe hearing loss. The relatively large size can also be utilized to house a relatively large size battery. On the other hand, CIC device 310C is almost invisible,

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while the room for the circuit and the battery is very limited. Generally, the selection of the device type is a compromise among considerations such as the user's degree of hearing loss (if applicable), ear canal size, desire of invisibility, desired longevity between battery replacements or recharges, and cost.

[0029] FIG. 4A is a block diagram showing one embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as the single ear-level device 110. The circuit is housed in an earmold configured for use as one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0030] Ear-level device 110 includes an external microphone 420, an ear canal microphone 440, an car canal speaker 470, a processor 460, a wireless transceiver 450, an antenna 455, a voice operated exchange (VOX) circuit 430, and a battery 480. External microphone 420 picks up ambient sound such that a hearing impaired person wearing ear-level device 110 is not "isolated" when communicating to remote device 190. Thus, ear-level device enables the hearing impaired person to talk to another person either directly (if the other person is nearby) or through a device such as a cell phone. Ear canal microphone 440 detects sound from the ear canal. The sound includes primarily speech of the person wearing ear-level device 110. Ear canal speaker 470 transmits sound received from remote device 190 and/or external microphone 420 to the same ear canal. In one embodiment, ear canal microphone 440 and ear canal speaker 470 are implemented as one physical device. Processor 460 converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 190, and converts the signals received from remote device 190 and/or external microphone 420 to a sound audible to the person wearing ear-level device 110. In one embodiment, processor 460 includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art In one embodiment, processor 460 includes a speech recognition module to enhance the audio signal received by and/or transmitted from ear-level device 110. Wireless transceiver 450 and antenna 455 form a telemetry interface supporting telemetry link 195 between ear-level device 110 and remote device 190. Wireless transceiver 450 includes a wireless transmitter and a wireless receiver. The wireless transmitter receives the electrical signal representing the sound picked up from the ear canal from processor 460 and transmits an outgoing wireless signal representing the same sound to remote device 190. The wireless receiver receives an incoming wireless signal representing a remote sound from remote device 190 and transmits an electrical signal representing the remote sound to processor 460. Telemetry link 195 provides for bi-directional communication allowing simultaneous signal transmission, in both directions, between ear-level device 110 and remote device 190. VOX circuit 430 activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 420, ear canal microphone 440, end wireless transceiver 450. In one embodiment, VOX circuit 430 includes a voice-controlled switch connecting between battery 480 and the portions of ear-level device 110 that are voice activated. The switch is driven by the sound detected by at least one of external microphone 420, ear canal microphone 440, and wireless transceiver 450. It is turned on when sound detected by external microphone 420 exceeds a predetermined threshold, when sound detected by ear canal microphone 440 exceeds another predetermined threshold, and/or when wireless transceiver 450 receives the incoming wireless signal representing the remote sound. In one embodiment, as discussed with respect to FIG. 4B below, VOX circuit 430 gates or attenuates one or more sounds detected by external microphone 420, ear canal microphone 440, and wireless transceiver 450, to eliminate or reduce echo and ringing caused by the loop formed by ear canal speaker 470 and ear canal microphone 440. Battery 480 supplies the power needed for the operation of ear-level device 110. In one embodiment, battery 480 is a rechargeable battery that can be recharged without being taken out from ear-level device 110.

[0031] FIG. 4B is a block diagram showing one embodiment of portions of VOX circuit 430. In this embodiment, VOX circuit 430 includes an ambient sound gating module 421 that blanks or attenuates the sound detected by external microphone 420, an occluded sound gating module 441 that blanks or attenuates the sound detected by ear canal microphone 440, a remote sound gating module 451 that blanks or attenuates the sound detected by wireless transceiver 450, and controller 431 to coordinate the states of the gating modules. Because ear canal microphone 440 is capable of detecting the sound delivered to the ear canal from ear canal speaker 470, there is a need to avoid echo and/or ringing. For example, a remote sound from remote device 190 delivered to the ear canal may be detected as an occluded sound and transmitted back to remote device 190, causing echo. The gating modules function in a coordinated manner to eliminate or reduce such echo and/or ringing caused by breaking, or applying resistance into, the audio loop formed by ear canal speaker 470 and ear canal microphone 440.

[0032] For the purpose of description, each of the gating modules is "on" when sound is "gated on" and "off' when the sound is "gated off" A sound is "gated on" when it is allowed to pass and be processed, and is "gated off' when it is blanked or substantially attenuated. Blanking refers to a substantially complete blockage of a sound, or in other words, that a detected sound is practically ignored by processor 460 such that it does not cause any echo or ringing that is audible by an ear. Substantial attenuation refers to an attenuation after which the attenuated sound does not cause any echo

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or tinging having an intolerable intensity.

[0033] Ambient sound gating module 421 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected ambient sound with a predetermined threshold for the ambient sound. When the amplitude of the detected ambient sound exceeds the threshold for the ambient sound, and when controller 431 permits, ambient sound gating module 421 is turned on. When the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off. In one embodiment, after the amplitude of the detected ambient sound drops below the threshold for the ambient sound, ambient sound gating module 421 is turned off after a predetermined delay.

[0034] Occluded sound gating module 441 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a detected occluded sound with a predetermined threshold for the occluded sound. When the amplitude of the detected occluded sound exceeds the threshold for the occluded sound, and when controller 431 permits, occluded sound gating module 441 is turned on. When the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off. In one embodiment, after the amplitude of the detected occluded sound drops below the threshold for the occluded sound, occluded sound gating module 441 is turned off after a predetermined delay.

[0035] In one embodiment, remote sound gating module 451 includes a voice-driven switch or attenuator driven by the output of a comparator that compares the amplitude of a remote sound with a predetermined threshold for the remote sound. When the amplitude of the detected remote sound exceeds the threshold for the remote sound, and when controller 431 permits, remote sound gating module 451 is turned on. When the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off. In one embodiment, after the amplitude of the detected remote sound drops below the threshold for the remote sound, remote sound gating module 451 is turned off after a predetermined delay.

[0036] In another embodiment, remote sound gating module 451 includes a switch or attenuator driven by the detection of the incoming wireless signal representing the remote sound. Remote sound gating module 451 is turned on upon detection of the incoming wireless signal when controller 431 permits, and turned off when the incoming wireless signal is no longer detected. In one embodiment, remote sound gating module 451 is turned off after the incoming wireless signal is absent for a predetermined period.

[0037] Controller 431 coordinates the on/off states of all the gating modules so the sounds are processed in an orderly fashion. Even if a sound is detected, the gating module corresponding to that sound is turned on only when controller 431 permits. In one embodiment, assuming that sounds A and B are gated by gating modules A and B, respectively, if both sounds A and B are detected, whichever sound is detected earlier is gated on. That is, if sound A exceeds its threshold first, gating module A is turned on and remain on until sound A drops below threshold A. Then, gating module B is turned only if sound B exceeds its threshold after gating module A has been turn off.

[0038] In one embodiment, controller 431 coordinates the on/off states of all the gating modules according to predetermined or programmed gating coordination rules. According to one exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows only the transmission of the remote sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. According to another exemplary rule, ambient sound gating module 421 is on, occluded sound gating module 441 is off, and remote sound gating module 451 is on. This rule allows the transmission of both the remote sound and the ambient sound to ear canal 303, and prevents the remote sound from being detected by ear canal microphone 440 and echoed back to remote device 190. This also prevents the ambient sound from being transmitted to remote device 190. According to yet another exemplary rule, ambient sound gating module 421 is off, occluded sound gating module 441 is on, and remote sound gating module 451 is off. When the person wearing ear-level device 110 speaks, both external microphone 420 and ear canal microphone 440 detect the voice. The voice as detected by ear-level device 110 is first transmitted to remote device 190. Without the gating modules (or when ambient sound gating module 421 and occluded sound gating module 441 are both on), the same voice as detected by external microphone 420 is transmitted to ear canal 303, and therefore again detected by ear canal microphone 440, and again be transmitted to remote device 190. This rule allows only the transmission of the occluded sound to remote device 190, and prevents the same sound from being transmitted to remote device 190 twice. Other rules are applied as a person skilled in the art should see fit based on an understanding after reading this entire document.

[0039] FIG. 5 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIGS. 3A, 3B, or 3C used as each of ear-level devices 210A and 210B. The circuit of each of ear-level devices 210A and 210B is housed in an earmold configured for use as of one of the device types illustrated as ear-level devices 310A, 310B, or 310C.

[0040] Ear-level device 210A is an audio transmitter that includes an ear canal microphone 540, a processor 560A, a wireless transmitter 550A, an antenna 555A, a transmitter VOX circuit 530A, and a battery 580A. Ear canal microphone 540 detects sound from the ear canal.

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The sound includes primarily speech of the person wearing ear-level device 210A. Processor 560A converts the sound picked up from the ear canal to an electrical signal to be transmitted to remote device 290. In one embodiment, processor 560A includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560A includes a speech recognition module used to enhance the sound transmitted from ear-level device 210A. Wireless transmitter 550A and antenna 555A form a telemetry interface supporting signal transmission from ear-level device 210A to remote device 290. Wireless transmitter 550A receives the electrical signal representing the sound picked up from the ear canal from processor 560A and transmits an outgoing wireless signal representing the same sound to remote device 290. VOX circuit 530A activates a major portion of ear-level device 210A only when sound is detected by ear canal microphone 540. In one embodiment, VOX circuit 530A includes a voice-controlled switch connecting between battery 580A and the portions of ear-level device 210A that are voice activated. The switch is driven by the sound detected by ear canal microphone 540. It is turned on when sound detected by ear canal microphone 540 exceeds a predetermined threshold. In one embodiment, VOX circuit 530A activates processor 560A when the sound is detected. In one further embodiment, VOX circuit 530A also activates wireless transmitter 550A when the sound is detected. Battery 580A supplies the power needed for the operation of ear-level device 210A. In one embodiment, battery 580A is a rechargeable battery that can be recharged without being taken out from ear-level device 210A.

[0041] Ear-level device 210B is an audio receiver that includes an external microphone 520, an ear canal speaker 570, a processor 560B, a wireless receiver 550B, an antenna 555B, a receiver VOX circuit 530B, and a battery 580B. External microphone 520 picks up ambient sound such that the person wearing ear-level devices 210A and 210B is not "isolated" when communicating to remote device 290. It allows that person to talk to a nearby person directly, without the need to take off ear-level devices 210A and/or 210B. Bar canal speaker 570 transmits sound received from remote device 290 and/or external microphone 520 to the same ear canal. Processor 560B converts the signals received from remote device 290 and/or external microphone 520 to a sound audible to the person wearing ear-level device 210B. In one embodiment, processor 560B includes one or more of amplification circuitry, filtering circuitry, acoustic feedback reduction circuitry, noise reduction circuitry, and tone control circuitry, among other circuits performing signal processing functions as known in the art. In one embodiment, processor 560B includes a speech recognition module used to enhance the audio signals received by ear-level device 210B. Wireless receiver 550B and antenna 555B form a telemetry interface supporting signal transmission from remote device 290 to ear-level device 210B. Wireless receiver 550B receives an incoming wireless signal representing a remote sound from remote device 290 and transmits an electrical signal representing the remote sound to processor 560B. VOX circuit 530B activates a major portion of ear-level device 110 only when sound is detected by at least one of external microphone 520 and wireless receiver 550B. In one embodiment, VOX circuit 530B includes a voice-controlled switch connecting between battery 580B and the portions of ear-level device 210B that are voice activated. The switch is driven by the sound detected by at least one of external microphone 520 and wireless receiver 550B. It is turned on when sound detected by external microphone 520 exceeds a predetermined threshold and/or when wireless transceiver 550B receives the incoming wireless signal representing the remote sound. In one embodiment, VOX circuit 530B activates processor 560B when the sound is detected. In one further embodiment, VOX circuit 530B also activates ear-level speaker 570 when the sound is detected. Battery 580B supplies the power needed for the operation of ear-level device 210B. In one embodiment, battery 580B is a rechargeable battery that can be recharged without being taken out from ear-level device 210B.

[0042] System 200 accommodates a larger overall circuit size without necessarily increasing the size of each ITE device. While two separate ear-level devices are required, because system 200 includes separate audio transmitter (ear-level device 210A) and receiver (ear-level device 210B) disposed in the ear canals of opposite ears, ear canal speaker 570 and ear canal microphone 540 do not form a loop causing echo and ringing.

[0043] FIG. 6 is an illustration of an exemplary exterior configuration of an ear-level device including a behind-the-ear (BTE) device 610 used as ear-level device 110 or one of ear-level devices 210A and 210B. BTE device 610 is attached to ear 101 with pinna 302 and ear canal 303. In one embodiment, BTE device 610 is used when none of ITE device 310A, PTC device 310B, and CIC device 310C provides sufficient space accommodating the required circuit and battery sizes. For the purpose of this document, only a device attached to a right ear is shown for each exterior structure. It is to be understood that each of ear-level devices 110, 210A, and 210B can be either a BTE device configured for right ear attachment or a BTE device configured for left ear attachment. Generally, a BTE device configured for right ear attachment has a substantially symmetrical appearance to a device of the same type configured for left ear attachment In system 100, ear-level device 110 is a device configured either for right ear attachment or for left ear attachment. In system 200, either one of ear-level device 210A and 210B is a device structured for right

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ear attachment, while the other is a device structured for left ear attachment.

[0044] BTE device 610 includes a behind-the-ear-module (BTE module) 611 and an in-the-ear module (ITE module) 612. ITE module 612 includes a custom or modular earmold. In one embodiment, the earmold is fully occluded to maximize audio efficiency. In another embodiment, the earmold is vented to remove undesirable low frequency sound.

[0045] FIG. 7 is a block diagram showing one embodiment of the circuit of ear-level device shown in FIG. 6 used as ear-level device 110. The exterior configuration of ear-level device 110 is of the type illustrated as BTE device 610.

[0046] Ear-level device 110 configured as BTE device 610 differs from ear-level device 110 configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 7 illustrates one embodiment in which BTE module 611 includes external microphone 420, processor 460, wireless transceiver 450, antenna 455, VOX circuit 430, and battery 480, and ITE module 612 includes ear canal microphone 440 and ear canal speaker 470. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, behind-ear portion 611 includes only battery 480 to maximize the duration of use of ear-level device 110 between battery replacements or recharges.

[0047] FIG. 8 is a block diagram showing another embodiment of the circuit of ear-level device shown in FIG. 6 used as each of ear-level devices 210A and 210B. The exterior configuration of each of ear-level devices 210A and 210B is of the type illustrated as BTE device 610.

[0048] Ear-level devices 210A/210B configured as BTE device 610 differ from ear-level devices 210A/210B configured as ITE/ITC/CIC devices 310A/310B/310C primarily in the physical arrangement of device components. By way of example, but not by way of limitation, FIG. 8 illustrates one embodiment of ear-level device 210A including a BTE module 611A and an ITE module 612A, and ear-level device 210B including a BTE module 611B and an ITE module 612B. BTE module 611A includes processor 560A, wireless transmitter 550A, antenna 555A, transmitter VOX circuit 530A, and battery 580A. ITE module 612A includes ear canal microphone 540. Behind-the-ear portion 611B includes external microphone 520, processor 560B, wireless receiver 550B, antenna 555B, receiver VOX circuit 530B, and battery 580B. ITE module 612B includes ear canal speaker 570. Other possible embodiments depend on size and other considerations that are known to those skilled in the art. In one embodiment, BTE modules 611A and 611B include only batteries 580A and 580B, respectively, to maximize the duration of use of ear-level devices 210A and 210B, respectively, between battery replacements or recharges.

[0049] FIG. 9 is a flow chart illustrating an embodiment of a method for audio communication using system 100 or system 200. The method provides for private, simultaneous two-way communication between a person and a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator. In this embodiment. VOX is used for power management. A VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 or system 200 that are voice activated. The VOX is "on" when the voice-activated portions of the circuit are activated, i.e., when the power switch remains on. It is "started" by turning on the power switch, and "stopped" by turning off the power switch.

[0050] Steps 910-918 illustrate the process of detecting a sound from an ear canal and transmitting it to a remote device. A microphone disposed in an ear canal detects an occluded sound within the ear canal at 910. If the occluded sound is detected at 911, the VOX of the system is started at 912 if it is not already on. The occluded sound is enhanced at 914 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice of a speech, and the noise is any sound other than the voice of the speech. The occluded sound is then converted to an outgoing radio signal at 916. The outgoing radio signal representing the occluded sound is transmitted to the remote device at 918.

[0051] Steps 920-928 illustrate the process of detecting a sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects an incoming radio signal representing a remote sound from the remote device at 920. If the incoming radio signal is detected at 921, the VOX of the system is started at 922 if it is not already on. The incoming radio signal is converted to a first incoming audio signal (the remote sound) at 924. The first incoming audio signal is enhanced at 926 with a voice recognition system that improves the signal-to-noise ratio, where the signal is the voice from the remote device, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the first incoming audio signal representing the sound from the remote device to the ear canal at 928.

[0052] Steps 930-938 illustrate the process of detecting a sound from the environment and transmit it to the ear canal. A microphone outside the ear canal detects an ambient sound at 930. If the ambient sound is detected at 931, the VOX of the system is started at 932 if it is not already on. The ambient sound is converted to a second incoming audio signal at 934. The second incoming audio signal is enhanced at 936 with a voice recognition system that improves the signal-to-noise ratio, where the signal is any voice included in the ambient sound, and the noise is any sound other than the voice. A speaker disposed in the ear canal transmits the second incoming audio signal representing to the ear canal at 928.

[0053] If no signal (sound or radio signal) is detected

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at 911, 921, and 931, the VOX is stopped at 940. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times. In one embodiment, the VOX is stopped after a predetermined period of time to avoid frequent activation/deactivation during a conversation.

[0054] Systems 100 and 200 are each a full duplex system capable of performing steps 910-918, 920-928, and 930-938 are performed in parallel and simultaneously. In an embodiment in which system 100 performs the method, ear-level device 110 performs steps 910-918, 924-928, and 930-938. If a signal is detected at 911, 921, or 931, VOX 430 activates at least portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively. If no signal is detected at 911, 921, or 931, VOX 430 deactivates at least the portions of device 110 performing steps 914-918, 924-928, or 934-938, respectively.

[0055] In an embodiment in which system 200 performs the method, ear-level device 210A performs steps 910-918, and ear-level device 210B performs steps 920-928 and 930-930. If a signal is detected at 911, transmitter VOX 530A activates at least portions of device 210A performing steps 914-918. If a signal is detected at 921 or 931, receiver VOX 530B activates at least portions of device 210B performing steps 924-928 or 934-938, respectively. If no signal is detected at 911, transmitter VOX 530A deactivates at least the portions of device 210A performing steps 914-918. If no signal is detected at 921 or 931, receiver VOX 530B deactivates at least the portions of device 210B performing steps 924-928 or 934-938, respectively.

[0056] FIG. 10 is a flow chart illustrating another embodiment of the method for audio communication using system 100. In this embodiment, in addition to power management, the VOX is also used for gating, i.e., blanking or attenuating, sounds. The VOX circuit includes a power switch connecting a battery to the portions of the circuit of system 100 that are voice activated and gating modules capable of blanking or attenuating the occluded sound, the remote sound, and the ambient sound. Examples of such gating modules include occluded sound gating module 441, remote sound gating module 451, and ambient sound gating module 421, designated for the occluded sound, the remote sound, and the ambient sound, respectively.

[0057] Steps 1010-1019 illustrate the process of detecting the occluded sound from the ear canal and transmitting it to the remote device. The microphone disposed in the ear canal detects the occluded sound within the ear canal at 1010. If the occluded sound is detected at 1011, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1012 if they are not already powered. The VOX determines whether to start gating the occluded sound on at 1013. In one embodiment, the occluded sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. Examples of gating coordination rules are described above with respect to FIG. 4B. If the occluded sound is gated on at 1013, it is processed at 1014 by following steps that are identical or similar to steps 914-916-918. The occluded sound is gated off at 1019 if it has been gated on but is no longer detected. In one embodiment, the occluded sound is gated off after it is not detected for a predetermined period.

[0058] Steps 1020-1028 illustrate the process of detecting the remote sound sent from the remote device and transmitting it to the ear canal. A radio receiver detects the incoming radio signal representing the remote sound from the remote device at 1020. If the remote sound is detected at 1021, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1022 if they are not already powered. The VOX determines whether to start gating the remote sound on at 1023. In one embodiment, the remote sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. In another embodiment, the remote sound is gated on when the incoming radio signal is detected and when the effective gating coordination rules permits. If the remote sound is gated on at 1023, it is processed at 1024 by following steps that are identical or similar to steps 924-926-928. The remote sound is gated off at 1029 if it has been gated on but is no longer detected. In one embodiment, the remote sound is gated off after it is not detected for a predetermined period.

[0059] Steps 1030-1038 illustrate the process of detecting the ambient sound and transmit it to the ear canal. A microphone outside the ear canal detects the ambient sound at 1030. If the ambient sound is detected at 1031, the VOX of the system is started to power the voice-activated portions of the circuit of system 100 at 1032 if they are not already powered. The VOX determines whether to start gating the ambient sound on at 1033. In one embodiment, the ambient sound is gated on when its amplitude exceeds a predetermined threshold and when the effective gating coordination rule permits. If the ambient sound is gated on at 1033, it is processed at 1034 by following steps that are identical or similar to steps 934-936-938. The ambient sound is gated off at 1039 if it has been gated on but is no longer detected. In one embodiment, the ambient sound is gated off after it is not detected for a predetermined period.

[0060] If no signal (sound or radio signal) is detected at 1011,1021, and 1031, the VOX is stopped at 1040. In other words, if no sound is to be processed, the communication system is inactivated except for the portions that must be active for sound detection at all times, In one embodiment, the VOX is stopped after a predetermined period of time to avoid Sequent activation/deactivation during a conversation.

[0061] It is to be understood that the above detailed description is intended to be illustrative, and not restric-

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tive. Other embodiments will be apparent to those of skill in the art upon reading and understanding the above description. For example, two ear-level devices 110, or a pair of devices each being similar to ear-level device 110, can be used to deliver a stereo sound. The scope of the invention should, therefore, be determined with reference to the appended claims.

Claims

1. A system adapted for use in at least one of a first ear having a first ear canal and a second ear having a second ear canal, comprising:

   a first ear-level device including:

   a first microphone adapted for use in one of the first ear canal and the second ear canal, the first microphone receiving an occluded sound from about the one of the first ear canal and the second ear canal;
   a first processor, connected to the first microphone, to process the occluded sound;
   a wireless transmitter, connected to the first processor, to receive the processed occluded sound from the first processor and transmit a first wireless signal representing the occluded sound; and
   a first earmold housing at least the first microphone, at least a portion of the first earmold configured for placement in the one of the first ear canal and the second ear canal.

2. The system according to claim 1, wherein the first earmold houses the first microphone, the first processor, and the wireless transmitter, and wherein the first earmold is configured for use as an in-the-ear (ITE) device.

3. The system according to claim 1, wherein the first ear level device comprises an ITE module including the first microphone and a behind-the-ear (BTE) module attached to the ITE module, the BTE module including the first processor and the wireless transmitter.

4. The system according to any of the preceding claims, further comprising a voice operated exchange (VOX) circuit connected to the first processor.

5. The system according to claim 4, wherein the VOX circuit comprises an occluded sound gating module, coupled to the first microphone, to blank or substantially attenuate the occluded sound.

6. The system according to claim 1, further comprising:

   a wireless receiver, connected to the first processor, to receive a second wireless signal representing a remote sound; and
   a speaker connected to the first processor, the speaker adapted for use in the one of the first ear canal and the second ear canal,

   wherein:

   the first processor further processes the remote sound, and the speaker is adapted to transmit the processed remote sound to the one of the first ear canal and the second ear canal; and
   the first earmold further houses at least the speaker.

7. The system according to claim 6, wherein the first microphone and the speaker include one common device receiving the occluded sound and transmitting the processed remote sound.

8. The system according to any of claims 6 and 7, further comprising a second microphone connected to the first processor, the second microphone adapted to receive an ambient sound from outside the one of the first ear canal and the second ear canal, and wherein:

   the first processor further processes the ambient sound; and
   the speaker is configured to transmit the processed remote and ambient sounds to the one of the first ear canal and the second ear canal.

9. The system according to claim 8, further comprising a voice operated exchange (VOX) circuit connected to the first processor.

10. The system according to claim 9, wherein the VOX circuit comprises an occluded sound gating module, coupled to the first microphone, to blank or substantially attenuate the occluded sound.

11. The system according to any of claims 9 and 10, wherein the VOX circuit comprises a remote sound gating module, coupled to the wireless transceiver, to blank or substantially attenuate the remote sound.

12. The system according to any of claims 9 -11, wherein the VOX circuit comprises an ambient sound gating module, coupled to the first microphone, to blank or substantially attenuate the ambient sound.

13. The system according to any of the preceding

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claims, wherein the first processor comprises a speech recognition module.

14. The system according to any of claims 6 -13, wherein the first earmold houses the first microphone, the first processor, the wireless transmitter, the wireless receiver, and the speaker, and wherein the first earmold is configured for use as an in-the-ear (ITE) device.

15. The system according to claim 14, wherein the first earmold is configured for use as an in-the-canal (ITC) device.

16. The system according to claim 15, wherein the first earmold is configured for use as a completely-in-the-canal (CIC) device.

17. The system according to any of claims 6 - 13, wherein the first ear level device comprises an ITE module including the first microphone and the speaker and a behind-the-ear (BTE) module attached to the ITE module and including the second microphone, the first processor, the wireless transmitter, and the wireless receiver.

18. The system of claim 17, wherein the BTE module comprises a voice operated exchange (VOX) circuit.

19. The system of claim 17, wherein the BTE module comprises a rechargeable battery.

20. The system according to any of claims 6 - 19, further comprising a remote device communicatively coupled to the wireless transmitter and the wireless receiver via a telemetry link providing for simultaneous bi-directional communication between the first ear-level device and the remote device.

21. The system according to claim 20, wherein the remote device comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

22. The system according to claim 1, wherein the first microphone is adapted for use in the first ear canal and receives the occluded sound from about the first ear canal, wherein the at least the portion of the first earmold is configured for placement in the first ear canal, and further comprising a second ear-level device including:

a wireless receiver to receive a second wireless signal representing a remote sound;
a second processor, connected to the wireless receiver, to convert the second wireless signal to the remote sound;
a speaker connected to the second processor, the speaker adapted for use in the second ear canal to transmit the remote sound to the second ear canal; and
a second earmold housing at least the speaker, at least a portion of the second earmold configured for placement in the second ear canal.

23. The system according to claim 22, wherein the second ear-level device further comprises a second microphone connected to the second processor, the second microphone adapted to receive an ambient sound from outside the second ear canal, and wherein:

the second processor further processes the ambient sound; and
the speaker is configured to transmit the remote sound and the processed ambient sound to the second ear canal.

24. The system according to any of claims 22 and 23, wherein the first ear-level device further comprises a first voice operated exchange (VOX) circuit connected to the first processor, and the second ear-level device further comprises a second VOX circuit connected to the second processor,

25. The system according to any of claims 22 - 24, wherein the first processor and the second processor each comprise a speech recognition module.

26. The system according to any of claims 22 - 25, wherein the first and second ear-level devices are each an behind-the-ear (BTE) device.

27. The system according to any of claims 22 - 25, wherein the first and second ear-level devices are each an in-the-ear (ITE) device.

28. The system according to claim 27, wherein the first and second ear-level devices are each an in-the-canal (ITC) device.

29. The system according to claim 28, wherein the first and second ear-level devices are each an completely-in-the-canal (CIC) device.

30. The system according to any of claims 22 - 29, further comprising a remote device communicatively coupled to the first and second ear-level devices.

31. The system according to claim 30, wherein the remote device comprises at least one of a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie, or a language translator.

32. A method, comprising:

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detecting an occluded sound using a first microphone adapted for use in an ear canal;
detecting an incoming radio signal representing a remote sound from the remote device;
starting a voice operated exchange (VOX) housed within an ear-level communication device when at least one of the occluded sound and the incoming radio Signal is detected if the VOX is not already on, the ear-level communication device being one of an in-the-ear (ITE) device and a behind-the-ear (BTB) device;
gating the occluded sound on, and gating the remote sound off, when the occluded sound is detected while the incoming radio signal is not detected; and
gating the remote sound on, and gating the occluded sound off, when the incoming radio signal is detected while the occluded sound is not detected.

33. The method according to claim 32, wherein detecting the occluded sound comprises comparing an amplitude of the occluded sound to an occluded sound threshold, wherein the occluded sound is detected when the amplitude of the occluded sound exceeds the occluded sound threshold.

34. The method according to any of claims 32 and 33, wherein detecting the incoming radio signal comprises comparing an amplitude of the remote sound to a remote sound threshold, wherein the incoming radio signal is detected when the amplitude of the remote sound exceeds the remote sound threshold.

35. The method according to any of claims 32 - 34, further comprising detecting an ambient sound using a second microphone, and wherein starting the VOX comprises starting the VOX when at least one of the occluded sound, the incoming radio signal, and the ambient sound is detected if the VOX is not already on.

36. The method according to claim 35, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound off, when the incoming radio signal is detected while the occluded sound is not detected.

37. The method according to claim 35, further comprising gating the occluded sound off, gating the remote sound on, and gating the ambient sound on, when the incoming radio signal is detected while the occluded sound is not detected.

38. The method according to claim 35, further comprising gating the occluded sound on, gating the remote sound off, and gating the ambient sound off, when the occluded sound is detected while the incoming radio signal is not detected.

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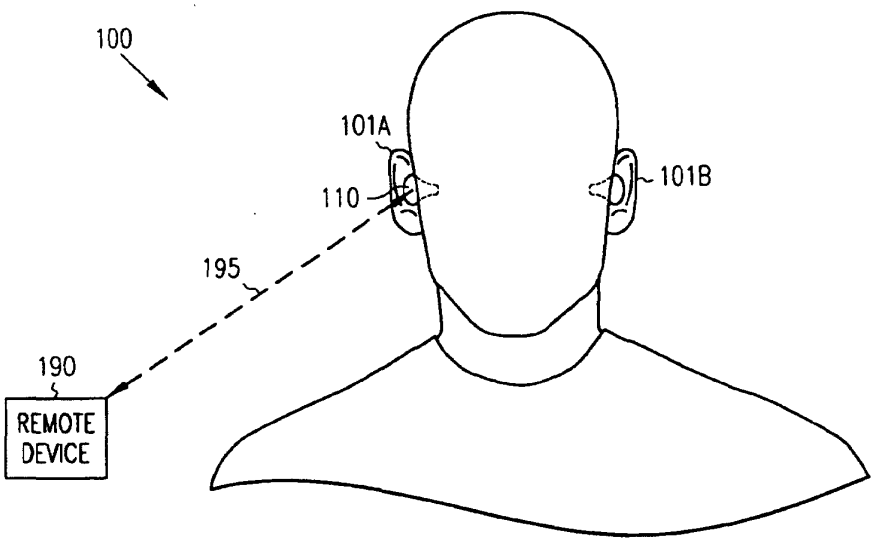

100
101A
101B
110
195
190
REMOTE DEVICE

FIG. 1

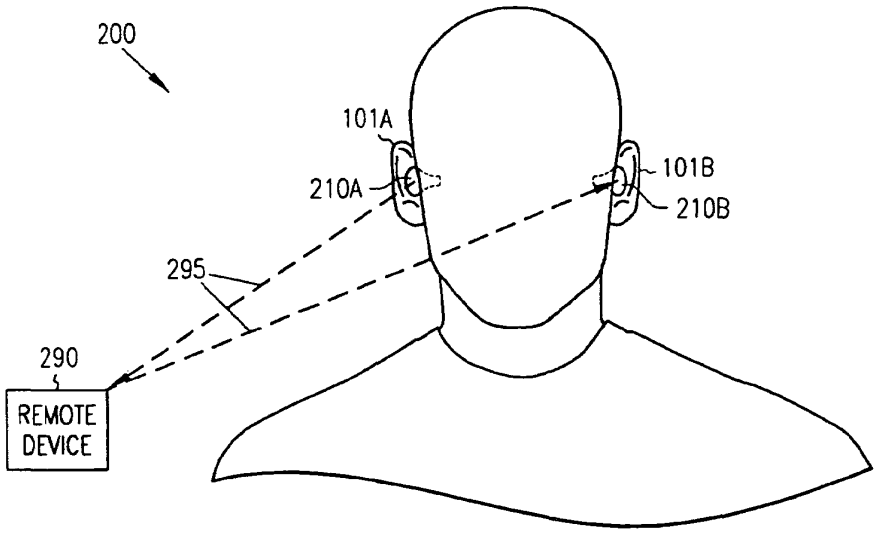

200
101A
101B
210A
210B
295
290
REMOTE DEVICE

FIG. 2

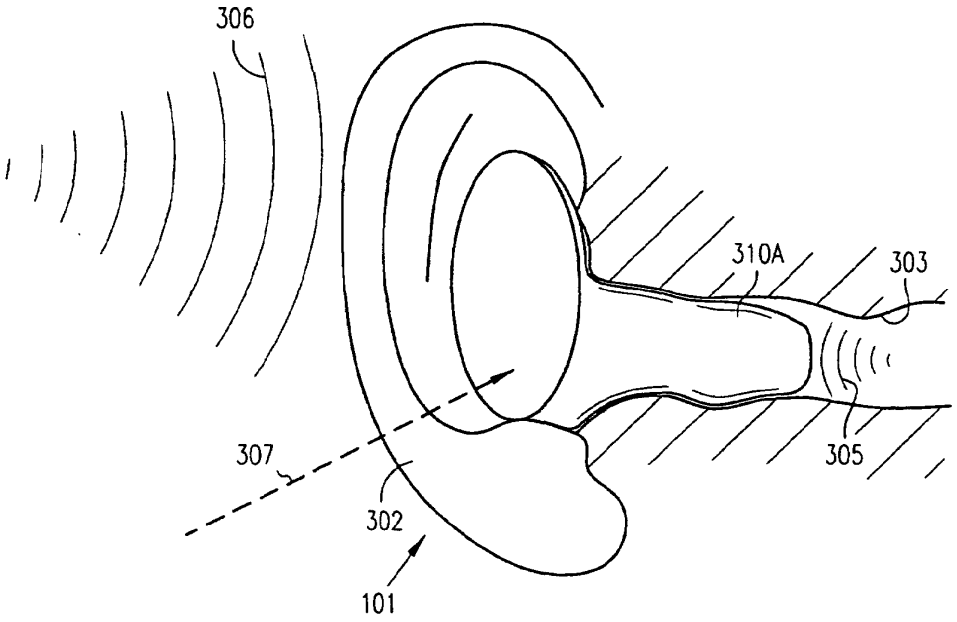


FIG. 3A

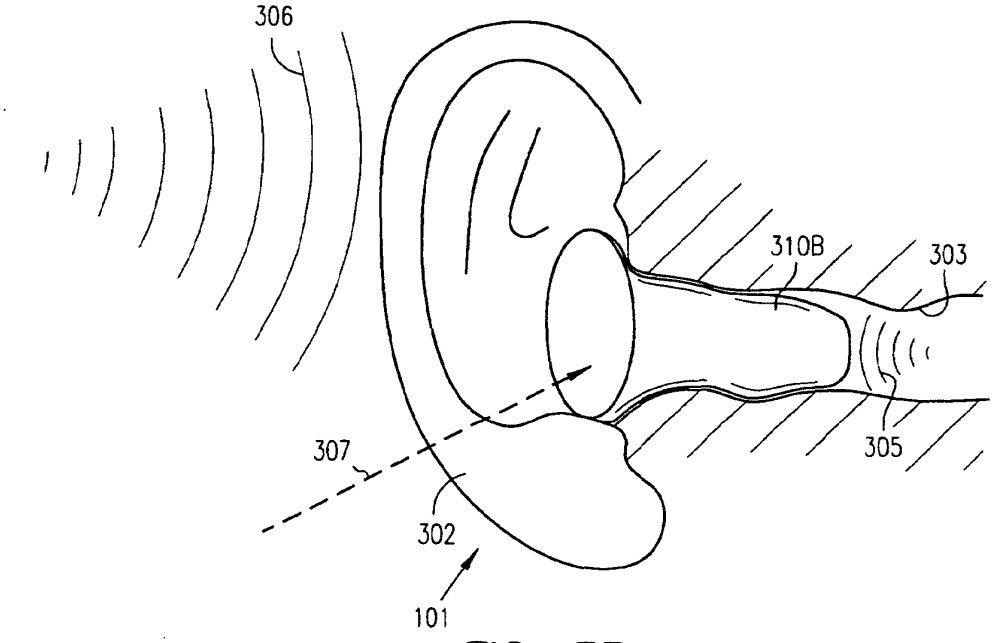


FIG. 3B

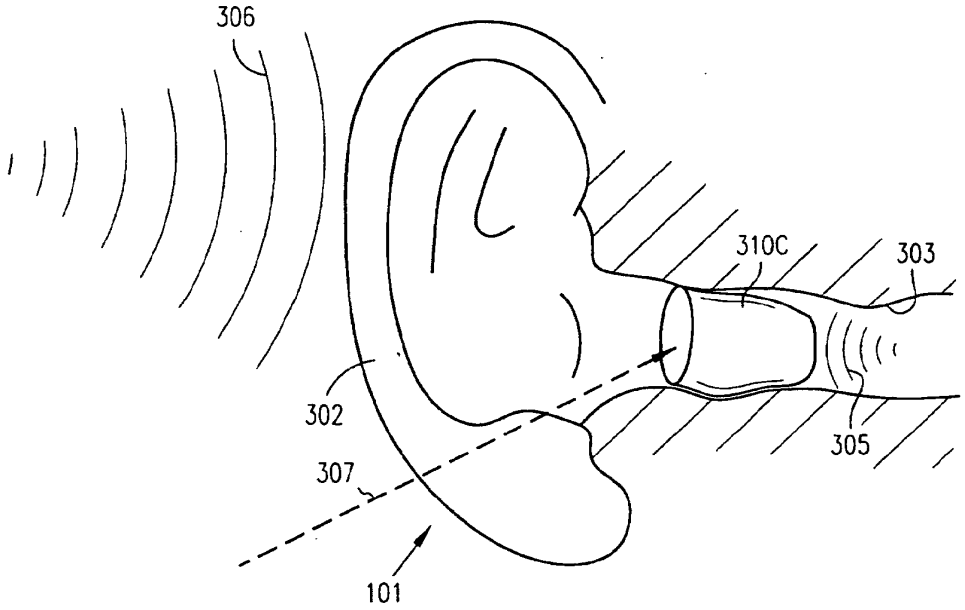


FIG. 3C

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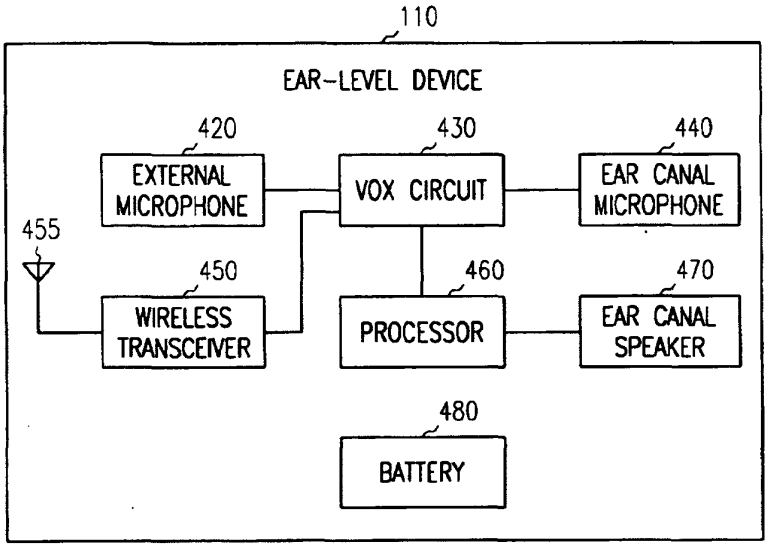

110
EAR-LEVEL DEVICE
420
EXTERNAL MICROPHONE
430
VOX CIRCUIT
440
EAR CANAL MICROPHONE
455
450
WIRELESS TRANSCEIVER
460
PROCESSOR
470
EAR CANAL SPEAKER
480
BATTERY

FIG. 4A

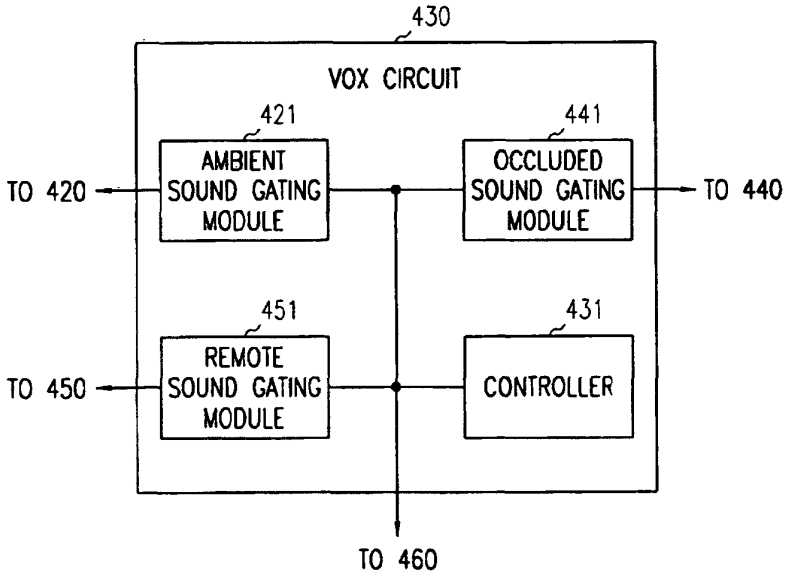

430
VOX CIRCUIT
421
AMBIENT SOUND GATING MODULE
441
OCCLUDED SOUND GATING MODULE
TO 420
TO 440
451
REMOTE SOUND GATING MODULE
431
CONTROLLER
TO 450
TO 460

FIG. 4B

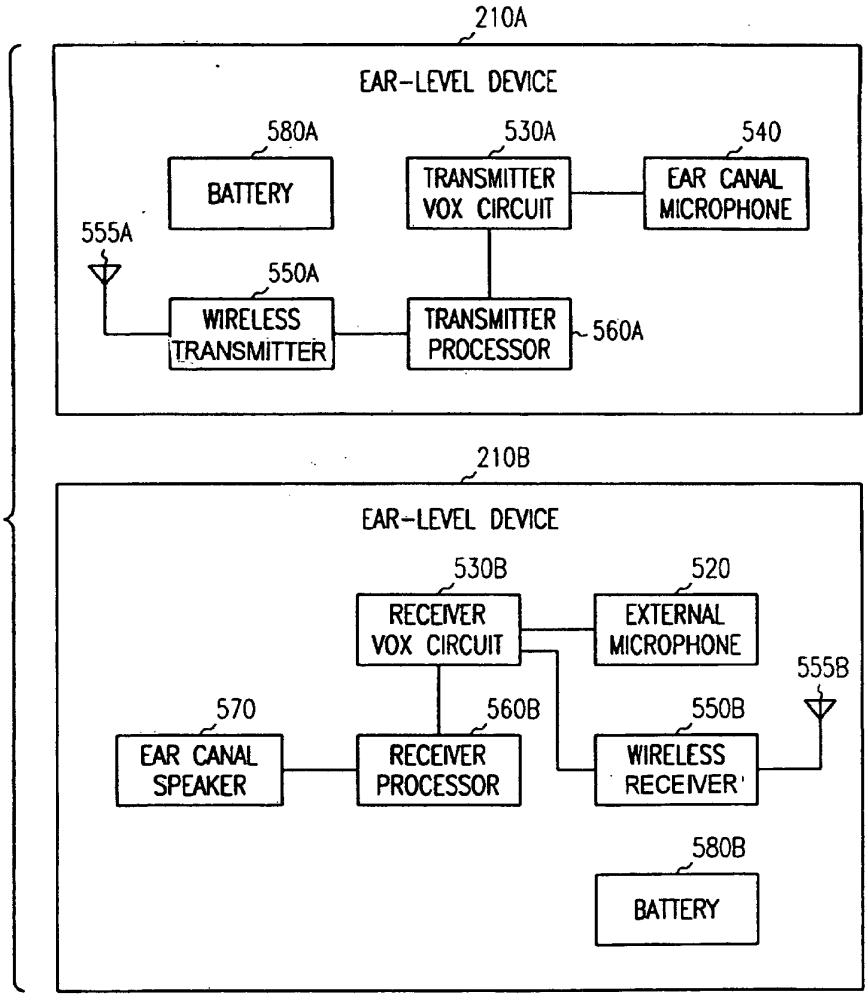

210A
EAR-LEVEL DEVICE
580A
BATTERY
530A
TRANSMITTER VOX CIRCUIT
540
EAR CANAL MICROPHONE
555A
550A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
560A
210B
EAR-LEVEL DEVICE
530B
RECEIVER VOX CIRCUIT
520
EXTERNAL MICROPHONE
555B
570
EAR CANAL SPEAKER
560B
RECEIVER PROCESSOR
550B
WIRELESS RECEIVER
580B
BATTERY

FIG. 5

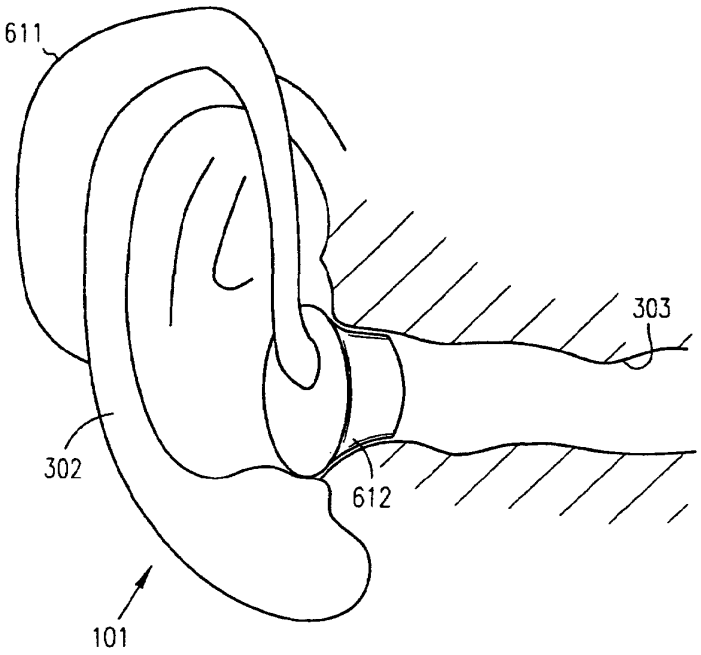

611
303
302
612
101

FIG. 6

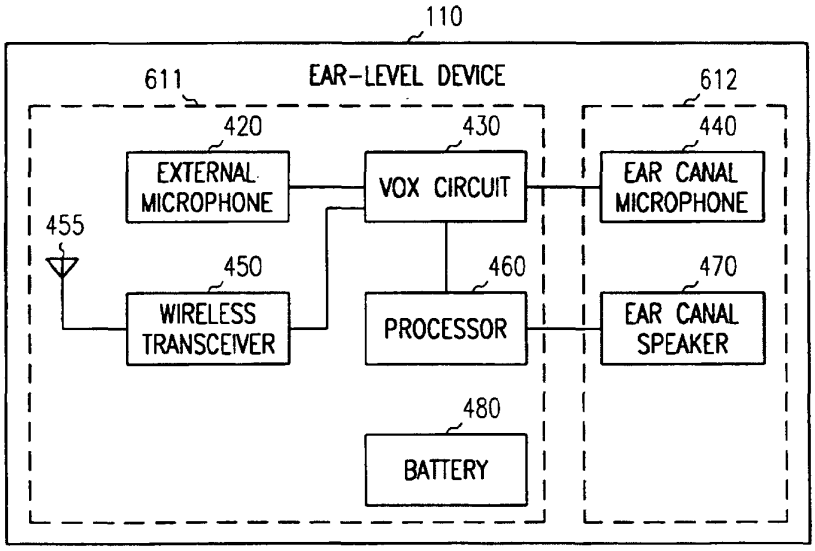

110
611
EAR-LEVEL DEVICE
612
420
430
440
EXTERNAL MICROPHONE
VOX CIRCUIT
EAR CANAL MICROPHONE
455
450
460
470
WIRELESS TRANSCEIVER
PROCESSOR
EAR CANAL SPEAKER
480
BATTERY

FIG. 7

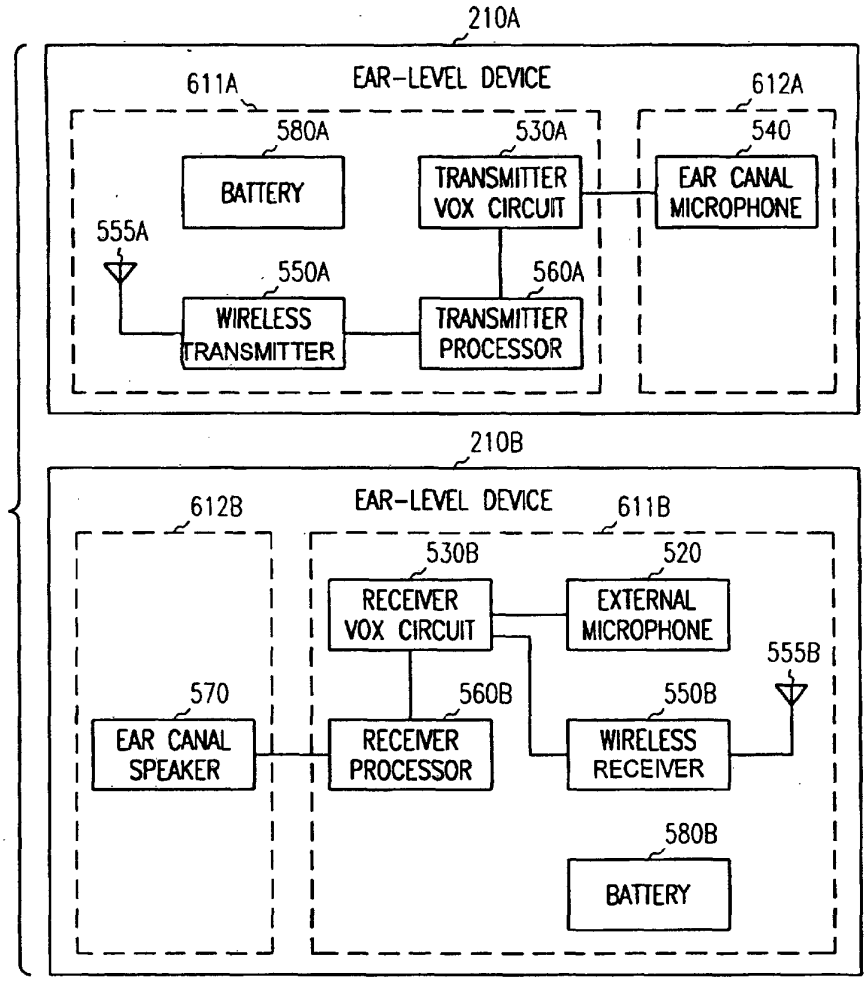

210A
611A
EAR-LEVEL DEVICE
612A
580A
530A
540
BATTERY
TRANSMITTER VOX CIRCUIT
EAR CANAL MICROPHONE
555A
550A
560A
WIRELESS TRANSMITTER
TRANSMITTER PROCESSOR
210B
612B
EAR-LEVEL DEVICE
611B
530B
520
RECEIVER VOX CIRCUIT
EXTERNAL MICROPHONE
555B
570
560B
550B
EAR CANAL SPEAKER
RECEIVER PROCESSOR
WIRELESS RECEIVER
580B
BATTERY

FIG. 8

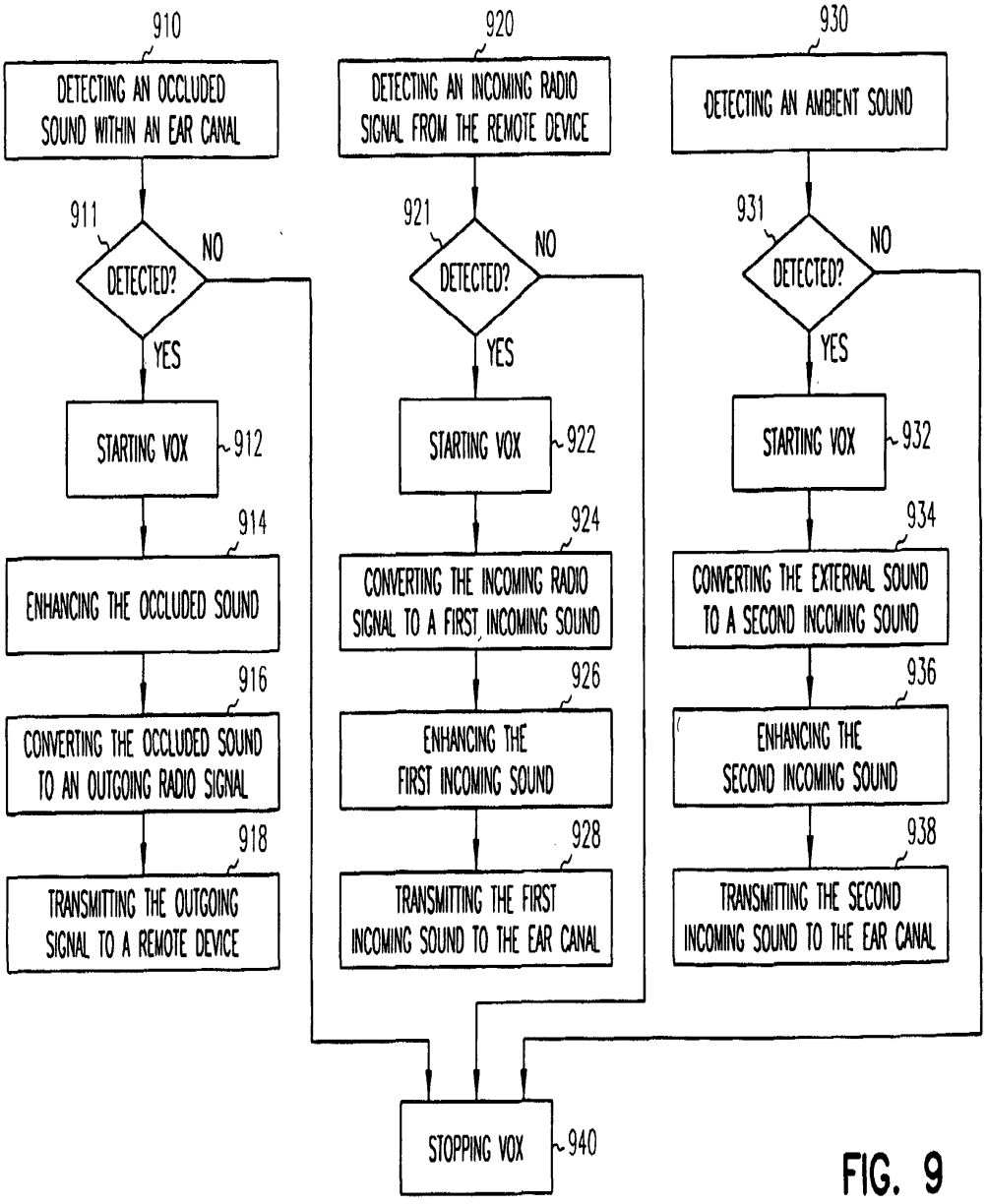

910
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
911
DETECTED?
NO
YES
STARTING VOX
912
914
ENHANCING THE OCCLUDED SOUND
916
CONVERTING THE OCCLUDED SOUND TO AN OUTGOING RADIO SIGNAL
918
TRANSMITTING THE OUTGOING SIGNAL TO A REMOTE DEVICE
920
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
921
DETECTED?
NO
YES
STARTING VOX
922
924
CONVERTING THE INCOMING RADIO SIGNAL TO A FIRST INCOMING SOUND
926
ENHANCING THE FIRST INCOMING SOUND
928
TRANSMITTING THE FIRST INCOMING SOUND TO THE EAR CANAL
930
DETECTING AN AMBIENT SOUND
931
DETECTED?
NO
YES
STARTING VOX
932
934
CONVERTING THE EXTERNAL SOUND TO A SECOND INCOMING SOUND
936
ENHANCING THE SECOND INCOMING SOUND
938
TRANSMITTING THE SECOND INCOMING SOUND TO THE EAR CANAL
STOPPING VOX
940

FIG. 9

20

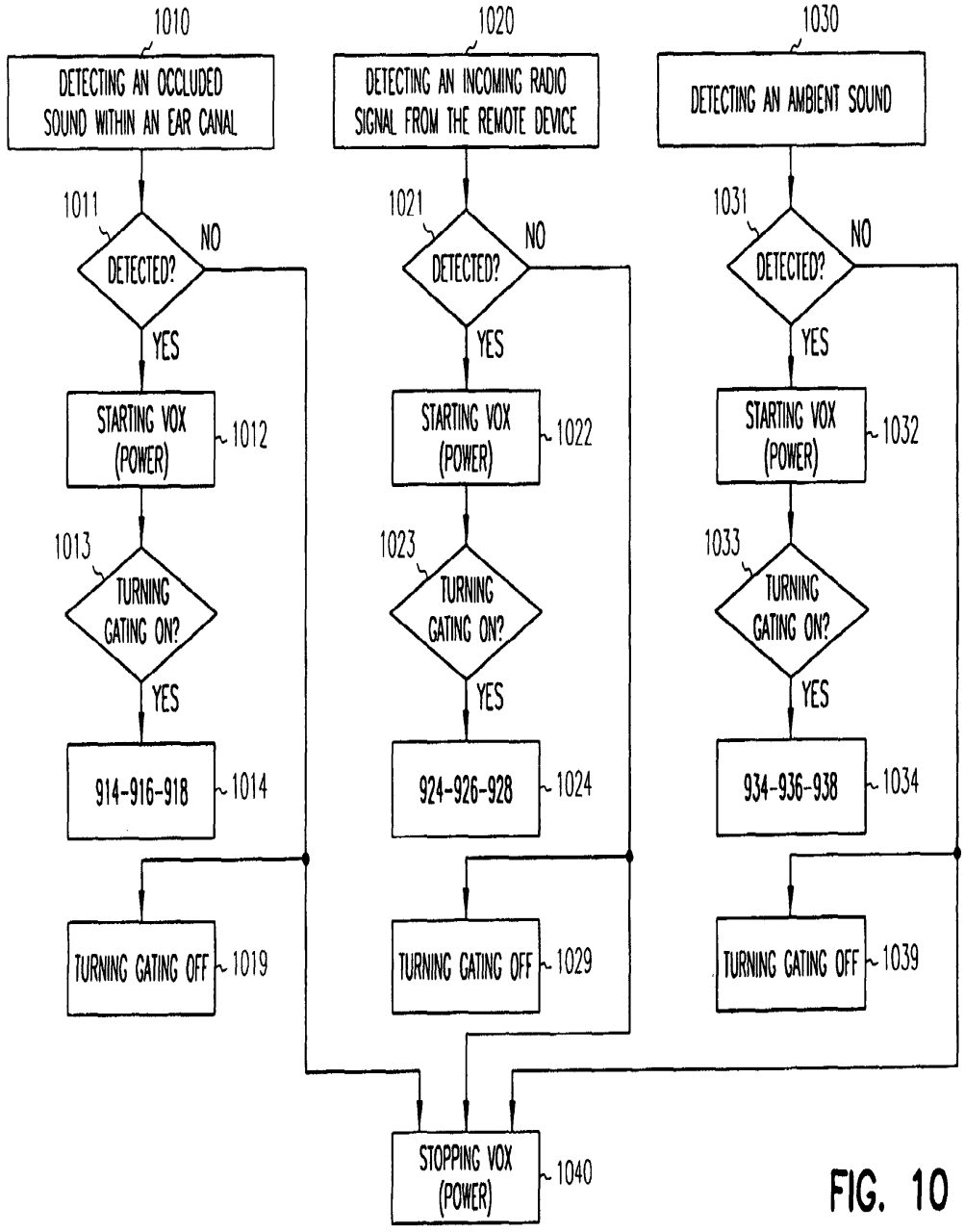

1010
DETECTING AN OCCLUDED SOUND WITHIN AN EAR CANAL
1011
DETECTED?
NO
YES
STARTING VOX (POWER)
1012
1013
TURNING GATING ON?
YES
914-916-918
1014
TURNING GATING OFF
1019
1020
DETECTING AN INCOMING RADIO SIGNAL FROM THE REMOTE DEVICE
1021
DETECTED?
NO
YES
STARTING VOX (POWER)
1022
1023
TURNING GATING ON?
YES
924-926-928
1024
TURNING GATING OFF
1029
1030
DETECTING AN AMBIENT SOUND
1031
DETECTED?
NO
YES
STARTING VOX (POWER)
1032
1033
TURNING GATING ON?
YES
934-936-938
1034
TURNING GATING OFF
1039
STOPPING VOX (POWER)
1040

FIG. 10

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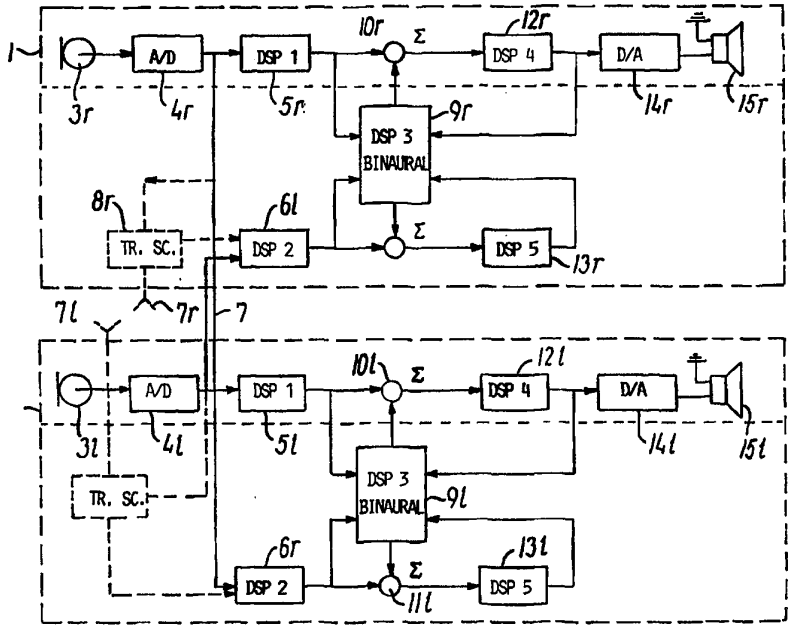
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(57) Abstract

A binaural digital hearing aid system comprises two hearing aid units (1, 2) for arrangement in a user's left and right ear, respectively. Each unit comprises input signal transducer means (3r, 3l), A/D conversion means (4r, 4l), digital signal processing means (5r–13r, 5l–13l), D/A conversion means (14r, 14l) and output signal transducer means (15r, 15l). A bi–directional communication link (7) is provided between the units. The digital signal processing means of each unit is arranged to affect a substantially full digital signal processing including individual processing of signals from the input transducer means of the actual unit and simulated processing of signals from the input transducer means of the other unit as well as binaural signal processing and includes at least a first digital signal processor part (5r, 5l) for processing said internally supplied signal, a second digital signal processor part (6l, 6r) for processing the signal supplied via said communication link (7) and a third digital signal processor part (9r, 9l) to effect common binaural digital signal processing of information derived from the signals processed in said first and second digital signal processor parts, said second digital signal processor part (6l, 6r) in each unit simulating the first digital signal processor part (5l, 5r) in the other unit with respect to adjustment parameters controlling the performance of said first signal processor part in said other unit.



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A Binaural Digital Hearing Aid System

The invention relates to a binaural digital hearing aid system comprising two hearing aid units for arrangement in a user's left and right ear, respectively, each of said units comprising input signal transducer means for conversion of a received input sound signal into an analog input signal, A/D conversion means for conversion of said analog input signal into a digital input signal, digital signal processing means for processing said digital input signal and generating a digital output signal, D/A conversion means for conversion of said digital output signal into an analog output signal and output signal transducer means for conversion of said analog output signal into an output sound signal perceivable to the user, a bidirectional communication link being provided between said units to connect a point in the signal path between the input signal transducer means and the digital signal processing means in one of said units with a corresponding point in the signal path between the input signal transducer means and the digital signal processing means of the other of said units.

For normally hearing persons the ability to localize sounds in space defined as binaural hearing ability is an important part of the sound perception. Typically the amplitude of sound received by the ipsilateral ear which is closer to the source of sound, is of greater amplitude than the sound received by the opposite contralateral ear. This difference in sound level, although often small by itself, is of great importance for a human being's perception of the direction of an incident sound.

In the human hearing system binaural sound perception results from a complicated signal processing of sounds arriving at the left and right side ears, in

which time/phase and frequency distribution of the sound plays a decisive role. Thus, time/phase differences and frequency enhancement are important for determining directions in the horizontal and vertical planes, respectively.

With conventional analog hearing aids persons suffering from a binaural hearing impairment, i.e. a hearing loss affecting both ears, the customary practice has been to use two separate hearing aids adjusted to compensate individually for the hearing loss of the respective ear for which the hearing aid is operative and compensation of the loss of binaural sound perception, although typically made even worse by the very use of a hearing aid in both ears, has in most cases by and large been ignored.

As a relatively simple compensation, it has been suggested for each of the two hearing aids of an analog system to use a microphone with a pronounced direction dependent characteristic to provide an analog signal the level of which changes, when the hearing aid is moved from a position pointing towards the sound source to other position with a minimum level, when the hearing aid points in a direction at right angles to the direction to the sound source.

In US-A-3,509,289 a different concept for compensation of binaural hearing loss in an analog hearing aid system is disclosed involving the use of cross-coupled AGC circuitry for maintaining and enhancing the interaural level difference between contralateral and ipsilateral incident sound. In this system, the gain of each of a first and second amplifying channel is varied inversely with the output of the other channel by separate AGC circuits which are cross-coupled to stabilize the system.

With the introduction of digital signal processing

in hearing aids a significant improvement of hearing aid performance has become possible and more advanced proposals for binaural hearing loss compensation have seen the light.

Thus, US-A-5,479,522 discloses a hearing enhancement system comprising in addition to two hearing aid devices for arrangement in the left and right side ears, respectively, a body-worn pack comprising a remote digital signal processor connected to each of the hearing aid device by a down-link and an up-link for interactive digital processing of the audio signals for each ear based on signals received from both hearing aid devices. The common binaural digital signal processing is predetermined and limited to attenuation of noise and narrowing of the sound field or adapting the signal level in the two channels. The signals supplied to the common binaural signal processing are not affected by the individual hearing loss compensation in the two channels.

In addition, this prior art system reduces the comfort by requiring a separate body-worn signal processor in a addition to the two hearing aid devices and the physical links between the common binaural processor and the two hearing aid devices in the form of radio communications make the system susceptible to distortion affecting the quality of sound reproduction.

In WO 97/14268, a binaural digital hearing aid system is disclosed in which the need for a separate body-worn remote control processor has been eliminated by the use of two hearing aid devices for arrangement in the left and right side ears, respectively, each of which incorporates a digital signal processor to which not only the unprocessed audio signal generated by the microphone in the same hearing aid device is supplied, but also the unprocessed audio signal generated by the

microphone in the opposite hearing aid device, the latter audio signal from each of the two devices being supplied to the respective opposite device through a bidirectional communication link.

This prior art system can be switched between distinct modes of either full binaural signal processing or performance as a conventional monaural hearing aid, which in one embodiment is done by giving the user the option of disabling the digital signal processor by either physically removing an external digital processing unit or by disabling a digital processor.

In the binaural processing mode of this prior art system no account is taken of the difference with respect to hearing loss and compensation between the two ears and, somewhat generalized, the system could be seen as an advanced digital substitute for the above-mentioned relatively simple binaural compensation using microphones with a pronounced direction dependent characteristic.

On this background, it is the object of the invention to provide an improved digital binaural hearing aid system in which the above-mentioned shortcomings of prior art systems have been eliminated to provide for a binaural signal processing, which for persons with a binaural hearing loss will restore binaural sound perception while taking into account the difference in hearing loss and compensation between the two ears.

According to the invention, a binaural digital hearing aid system as defined above is characterized in that the digital signal processing means of each unit is arranged to effect a substantially full digital signal processing including individual processing of signals from the input transducer means of the actual unit and simulated processing of signals from the input transducer means of the other unit as well as binaural

signal processing of signals supplied, on one hand, internally from the input signal transducer means of the same unit and, on the other hand, via said communication link from the input signal transducer means of the other unit, said digital signal processing means including at least a first digital signal processor part for processing said internally supplied signal, a second digital signal processor part for processing the signal supplied via said communication link and a third digital signal processor part to effect common binaural digital signal processing of information derived from the signals processed in said first and second digital signal processor parts, said second digital signal processor part in each unit simulating the first digital signal processor part in the other unit with respect to adjustment parameters controlling the performance of said first signal processor part in said other unit.

Thereby, in the binaural hearing aid system of the invention each of the hearing aid units for the left and right side ears, respectively, performs in addition to digital signal processing adapted to compensate for the hearing loss of the ear served by the unit, a simulated full digital signal processing of sound signals received by the unit for the opposite ear and adapted to compensate for the specific hearing loss of that ear, as well as a common binaural signal processing taking into account both of the normally different compensation characteristics of both units.

By the advantageous embodiments and modifications of the system set out in the dependant claims the system can be designed for user operated switching between functioning as a binaural system and a conventional monaural hearing system, and the digital signal processing means in each hearing unit may be

programmable to be switchable between different sound environments or listening situations by user operation, whereby programmed performance data for the first signal processing means of one unit is entered for programming of the second signal processing mens of the other unit, in which the simulated signal processing of signals supplied from the first unit is carried out.

By the provision of only a single bidirectional communication link between the two hearing aid units, the hearing aid system of the invention is less susceptible to signal distortion and interruption than the prior art systems described above.

In the following the invention will be further explained with reference to the accompanying drawings on which

figure 1 is a schematic block diagram representation of an embodiment of the hearing aid system of the invention;

figure 2 a further detailed representation of the embodiment shown in figure 1; and

fig. 3 is a block diagram representation of one hearing aid unit of a programmable hearing aid system of the invention.

The binaural hearing system illustrated in figure 1 comprises two hearing aid units 1 and 2 intended for arrangement in the user's right and left ears, respectively. The hearing aid units 1 and 2 are identical in structure, but as further explained in the following they will normally have been programmed or otherwise adjusted to provide different hearing loss compensation adapted to the specific hearing impairment of the ear, in which the unit is to be arranged. For the following description identical parts in the two units 1 and 2 will be designated by the same reference numeral followed by "r" and "l", respectively, to indicate the

localization of such parts in either the right ear or the left ear unit 1 or 2.

Each of units 1 and 2 comprises input signal transducer means e.g. in the form of one or more hearing aid microphones 3r, 3l which receives sound signals to be processed in the unit and transforms these sound signals into analog electrical signals which are supplied to an A/D converter 4r, 4l for conversion into digital signals.

In the embodiment shown the digital signal from A/D converter 4r, 4l in each of units 1 and 2 is supplied to a first digital signal processor 5r, 5l which is programmed or otherwise adjusted to perform signal processing functions such as filtering, band-division, amplification, gain control adjustment, compression, expansion and/or compensation for unlinearities in the microphone or the user's ear channel.

However, to the extent that some of the signal processing functions of processors 5r, 5l may be implemented in the A/D converters 4r, 4l, each of which will then supply a preprocessed digital signal, each of digital processors 5r, 5l need not be present as a separate unit.

According to the invention, each of units 1 and 2 also comprises a second digital signal processor 6l and 6r, respectively, which is structurally identical to processor 5r, 5l, but is programmed or otherwise adjusted to perform digital signal processing functions on the signals received by the opposite ear, i.e. processor 6l in unit 1 for the right ear is programmed to provide the specific signal processing intended for the left ear and will thus, in principle, provide the same signal processing as signal processor 5l in unit 2, whereas signal processor 6r in unit 2 will provide the same signal processing as processor 5r in unit 1.

In the embodiment shown the digital electrical signal from converter 4r, 4l in each of units 1 and 2 is additionally supplied via a communication link 7 to second signal processor 6r, 6l in the other unit, such that in each unit the second signal processor 6l, 6r performs a simulated processing corresponding to the processing by first signal processor 5l, 5r in the other unit. However, as an alternative the analog signals from microphone 3r, 3l in each of units 1 and 2 could be communicated directly and supplied to A/D conversion in the other unit.

The signal processors 5r, 5l and 6r, 6l will typically be state of the art digital hearing aid processors programmed to perform a relatively sophisticated signal processing with respect to sound/noise separation and user operated adaptability to a number of different sound environments or listening situations.

The communication link 7 between the right and left hearing aid units 1 and 2 is preferably a single bidirectional communication link which may be physically implemented by a cable extending between the two units. the hearing aid units 1 and 2 may be designed for arrangement in the ear (ITE) or behind the ear (BTE). In either case a cable connection between the units may extend around the neck of the user and may eventually be integrated in a necklace or similar piece of jewelry or bijouterie.

Alternatively, the bidirectional communication link 7 may be wireless and, as shown in dashed lines,comprise antennas 7r, 7l connected with appropriate transceiving means 8r, 8l in each unit. For hearing units in ITE design such an antenna may be physically implemented by a relatively short piece of wire or string which in use will project outside the ear and

may serve additionally to facilitate withdrawal of the hearing unit from its ITE position.

In each of units 1 and 2 the first and second digital signal processors 5r, 5l and 6l, 6r outputs a processed digital signal which is supplied to a third signal processor 9r, 9l which, in accordance with the invention performs a common binaural digital signal processing of the processed digital signals outputted from the first and second signal processors 5r, 5l and 6l, 6r.

The binaural signal processing in each of third signal processors 9r, 9l may make use of state of the art binaural processing techniques taking into account differences with respect to amplitude, phase-lag etc. between arrival of incoming sounds at the input transducers of the right and left ear hearing aid units. As result of this binaural signal processing which according to the invention is based on information derived from the processed digital signals in both of units 1 and 2, the third signal processor 9l, 9r in each unit outputs processed digital right and left binaural signal parts which in digital adder devices 10r, 10l and 11l, 11r is combined with the processed digital output signal from first signal processor 5r, 5l in the same unit.

In each unit the combined processed digital signals from adder device 10r, 10l may be supplied directly to a D/A converter 14r, 14l for conversion into a processed analog signal which is supplied to an output transducer device in the form of a conventional hearing aid telephone 15r, 15l. As illustrated the processed digital signals from adder devices 10r, 10l and 11r, 11l may optionally undergo a further digital signal processing in fourth signal processors 12r, 12l and fifth signal processors 13r, 13l, respectively,

which may include compensation of the specific hearing loss and automatic gain control. From processors 12r, 12l and 13r, 13l feedback signals are also supplied to the binaural processor 9r, 9l.

The processing functions of the fourth and fifth signal processors 12r, 12l and 13r, 13l in each of units 1 and 2 may be implemented in the binaural processor 9r, 9l so that processors 12r, 12l and 13r, 13l may in principle be dispensed with as separate units. The binaural signal processor 9r, 9l may then further be designed to output only the binaural digital signal part intended for the actual unit, i,e, the right ear binaural signal part for unit 1 and the left ear binaural signal part for unit 2. In both cases, the incorporation of the fourth and fifth signal processors either as separate units or in the binaural processor 9r, 9l provides an advantageous possibility, however, for an AGC function and/or hearing loss compensation in the binaural signal processor 9r, 9l by feed-back of the processed digital output signals from both of the fourth and fifth signal processors 12r, 12l and 13r, 13l to the binaural processor 9r, 9l.

Examples of digital signal processors for use in each of units 1 and 2 are disclosed e.g. in EP-B1-0 732 036, US-A-5,165,017, US-A-4,531,229 and US-A-5,144,675. An advanced signal processing method and device employing socalled dynamic AGC has been disclosed in copending international patent application PCT/DK97/00598, the disclosure of which is incorporated herein by reference.

The design and structure of the binaural hearing system of the invention, by which all information carrying signals in the separate signal channels for the right ear and the left ear sound perception are made available for processing in both of units 1 and 2

not only of the signal actually belonging to the respective side, but also, by simulated processing, of the signal belonging to the other side, opens the possibility of implementing complex and highly sophisticated binaural signal processing to restore binaural hearing ability without significant complication of the structure. In fact, both of hearing aid units 1 and 2 may be identical in structure and equipped with identical components like converters, signal processors etc.

Whereas the various signal processors in each of units 1 and 2 have been illustrated and described as separate processors they may advantageously be incorporated as separate processing parts of a common single digital processor such as a microprocessor.

The embodiment of the hearing aid system shown in figure 2 serves to illustrate the degree of complexity of binaural signal processing that can be implemented in each of the right and left ear hearing aid units 16 and 17 with a single bidirectional communication link 28 between the two units.

Using the same distinction as to reference numerals between the right and left ear units as used for the embodiment in figure 1 only the structure and function of the right ear unit 16 will be explained in the following.

The unprocessed analog signal from microphone 18r is preamplified and converted to digital form in preamplifier and A/D converter 19r and processed to compensate for unlinearity of the microphone and the sound perception in the ear in linearity control unit 20r, from which a preprocessed digital signal is supplied, on one hand to a band divider filtering unit 21r in the signal processing channel for the right ear and, on the other hand, via the bidirectional communication link 28 to a band divider filtering unit 21rs in

the processing part of the left ear unit 17 performing the simulated right ear signal processing.

In the band divider filtering unit 21r the incoming preprocessed digital signal is split into a number of frequency bands, each of which is further processed in a noise filtering unit 22r and a processing unit 23r in which the signal is amplified in accordance with the compensation characteristic adapted to compensate for the specific hearing loss of the right ear.

As for the embodiment in figure 1 each of the two hearing aids comprise in addition to the signal processing channel for the ear in which the unit is arranged a separate signal processing channel effecting a simulated signal processing corresponding to the signal processing in the other unit. In the embodiment in figure 2 this simulated processing channel comprises for the right ear hearing aid unit 16 processing units 21ls, 22ls and 23ls effecting the same processing functions as processing units 21r, 22r and 23r for the right ear compensation, but adjusted to the specific characteristics for the left ear compensation effected in the left ear hearing aid unit 17.

The left ear hearing aid unit 17 is identical in structure to the right ear hearing aid unit 16 and comprises the left ear signal processing channel with processing unit 19l and 20l and filtering and compensation units 21l, 22l and 23 l as well as the simulated right ear processing channel including units 21rs, 22rs and 23rs.

In each of hearing aid units 16 and 17 binaural signal processing may be effected in two processing units 24, 24l and 25r, 25l. Thus, in the right ear hearing aid unit 16 a first binaural processing unit 24r may receive the band divided output signals from filtering unit 21r in the right ear processing channel

as well as from filtering unit 21ls in the simulated left ear processing channel and provide correction signals affecting signal scaling in processing units 22r and 22ls and a second binaural processing unit 25r may effect further binaural signal processing on incoming signal from the first binaural processing unit 24r as well as from processing units 22r, 22ls and 23r, 23ls.

Finally, in each of hearing aid units 16 and 17 the output signal from processing unit 23r, 23l in the right and left ear processing channel, respectively, and the binaural output signal from the second binaural processing unit 25r. 25l is reconverted into analog form in an output processing unit 26r, 26l and supplied to an output transducer such as a conventional hearing aid telephone 27r, 27l.

In each of hearing aid units 16 and 17 all of processing units 22 to 25 may be designed for automatic gain control (AGC), e.g. as disclosed in the above-mentioned copending international patent application PCT/DK97/000598.

In each of hearing aid units 16 and 17 the processing units 21 to 25 are thus interconnected via a multiplicity of internal information and control signals lines, whereas the only external connection to the other hearing aid unit is via the single directional communication link 28.

According to a particular aspect of the invention the signal lines connecting the processing units 21 to 23 of the right and left ear processing channels and the simulated left and right ear processing channels to the binaural processing units 24 and 25 may be opened and closed or activated and deactivated by control of appropriate switching means, not illustrated, whereby an advantageous adjustment flexibility is obtained with

a smooth transition ranging from full binaural signal processing approximating the sound information processing of the human brain via a more simple binaural sound level control to conventional monaural sound reproduction, contrary to the separation of the processing units for normal and binaural processing in the prior art system of WO 97/14268 explained above.

The signals supplied in each of the right and left ear hearing aid units to the binaural signal processing in units 24 and 25 may as illustrated in figure 2 be microphone signals which have been preprocessed by conversion into digital form and correction of frequency and/or level distortions caused by unlinearities in the microphone circuits and/or following from the arrangement in the user's ear channel. Preferably, the incoming signals for the binaural processing have been filtered to the desired frequency band width.

Moreover as shown in figure 2 the preprocessed microphone signal supplied from each of hearing aid units 16 and 17 to the simulated processing channel of the other unit may be limited, e.g. by compression in an additional compressor unit 28r, 28l, to reduce the dynamic range, the band width and/or the number of samplings, thereby reducing the amount of data or information to be processed by the simulated processing. A similar signal limitation may also be provided, e.g. by an additional compressor unit 29r, 29l for the signals supplied in each hearing unit from the signal processing channel for the right or left ear, respectively, to binaural signal processing. In either case the processing units, to which such compressed signals are supplied, must be designed for processing these signals.

The binaural signal processing effected by processing units 24 and 25 may comprise a level correc-

tion, by which the gain in the hearing aid unit, right or left, receiving the weakest incoming sound signal is controlled on the basis of the incoming sound signal at the other hearing aid unit as represented, e.g. by the preprocessed microphone signal communicated therefrom via communication link 28 for simulated signal processing. Thereby, the sound level ratio between sounds received by the right and left ears, respectively. and the spatial information provided thereby may be maintained also for hearing aid units with automatic gain control (AGC), sin AGC control can be effected on the basis of the strongest processed signal in the right or left ear units 16 and 17

The complexity allowed for the internal signal processing in each of hearing aid units 16 and 17 would also allow a compensation for time delays introduced e.g. by the signal communication via communication link 28, if necessary.

For hearing aids which as shown in figure 2 employs sound or signal processing in a number of separate frequency bands with automatic gain control in each band the processing complexity and/or capacity further entails a data or information transfer between the real and simulated processing channels in each unit to provide for equal adaptation of the gain control of these processing channels, whereby the overall transfer function of each hearing aid unit may be adapted to take account of sound spectrum differences occurring at the right and left sides, thereby taking account of the frequency distribution in the spectra of sounds received at the right and left ears, which is very important for the localization of a sound source in space.

Since, in each of the right and left ear hearing aid units 16 and 17 in figure 2 all processing units

19r - 23r, 19l - 23l for the actual right or left ear signal processing channel and all processing units 21ls - 23ls, 21rs - 23rs for the simulated signal processing channel are programmed or otherwise adjusted to the specific processing parameters for the right and left ear signal processing, respectively, or vice versa, the binaural signal processing in each side takes fully account of the specific hearing loss characteristics of both sides up to or even beyond the output signal transducer 26r and 26l, respectively.

To accomplish this the binaural signal processing effected in the right and left ear hearing units 16 and 17 will typically be mirror images of each other to restore the actual sound level and sound spectrum differences between incoming sounds at the right and left ears, respectively.

As a special advantage of the binaural signal processing in each of the two hearing units of the system according to the invention a sophisticated noise or feedback suppression is made possible, by which tone signals deviating from the overall sound image may be effectively suppressed without suppression of tone signals present in the overall sound signal or in the right and left side at the same time. This can be accomplished by including in the binaural sound processing unit a feedback suppression system to which a residual feedback signal representing the difference feedback signals from the actual and simulated sound processing channels is supplied. By means of such a feedback suppression it is possible for the hearing aid system of the invention to distinguish between howl and information sound signals of a similar character such as a tone from a flute solo in classical music composition or alarm or signalling tones such as walk/stop beeps at traffic lights.

In each of hearing aid units 16 and 17 the performance of each of signal processing units 21r - 23r, 21l - 23l in the real signal processing channel as well as the performance of each of the processing units 21ls - 23ls, 21rs - 23rs in the simulated processing channel is controlled by adjustment parameters or data adapted to the specific compensation requirements of the right and left ears, respectively.

According to the invention such adjustment parameters may be individually programmable to compensate for the user's specific hearing impairment with respect to the right and left ears, whereby the hearing aid system may be supplied with a standard adjustment to permit individual programming to be effected by a hearing aid fitter as is customary practice in the individual user adjustment of hearing aids.

Moreover, the adjustment parameters may be organized in different programme settings to permit operation of the hearing aid system in different modes ranging from fully binaural to simple monaural operation of the hearing aid units and/or permit adaption of the hearing aid system to varying sound environments or listening situations.

Figure 3 shows for one of the hearing aid units in the system illustrated in figure 2, i.e. the right ear hearing aid unit 16 how this is accomplished by means of a performance and programme memory 30 in which all programmable adjustment parameters for a number of specific performance programmes are entered and may be selected from a selection unit 31 which may be user operated and/or operated from a sound signal analyzer 32 to effect programme selection automatically in response to occurrence of specified sound signal conditions.

Optionally, at least one of hearing aid units 16

or 17 may include means for calculation of intermediate settings between at least two consecutive performance programme settings, in which case also such intermediate settings will be selectable from the selection unit 31.

Since as a result of the structure and organiza - tion of hearing aid units 16 and 17 to effect not only the actual signal processing for the ear in which the unit is arranged, but also the simulated processing for the opposite ear memory 30 will contain all adjustment parameters needed for the signal processing for both sides, the programming of the hearing aid system may be effected by entering of adjustment parameters and user operated or automatically activated performance programmes in memory 30 of the one of hearing aid units 16 and 17 only and effecting transfer of adjustment parameters for the processing units of the other hearing aid unit via the communication link 28 in an adjustment or initiation mode activated at each change of performance programme.

Ultimately this makes possible to operate the system according to the invention by a master-slave principle, in which case one of the hearing aid units would function as a master unit and take control of the other unit functioning as a slave unit in which memory 30 would then contain the information or parameters needed for the actual function of the slave unit.

Alternatively, various kinds of intermediate or mixed organization schemes could be foreseen, e.g. by designing both hearing units with user operated as well as automatic programme selection. This could provide e.g. for consensus operation in situations where one unit would try to shift automatically to a specific programme matching prevailing sound signal conditions, by effecting an exchange of actual adjustment parameter

settings between the two units via communication link 28 to enable a decision to be made in one of the units as to whether the programme selected by one of the units should be effected for both units.

In each of the two units synchronization means 33 may further be provided for the exchange of synchronization information between the signal processing parts of the two units via the communication link 28. Such synchronizing information may be derived from the signals otherwise transferred between the two units or be generated as separate synchronizing signals.

User operability may advantageously be effected by wireless remote control from a separate control unit carried by the user. This is suitable, in particular, for embodiments in which wireless transmission is already used for the bidirectional communication link between the two hearing aid units.

P A T E N T C L A I M S

1. A binaural digital hearing aid system comprising two hearing aid units for arrangement in a user's left and right ear, respectively, each of said units comprising input signal transducer means for conversion of a received input sound signal into an analog input signal, A/D conversion means for conversion of said analog input signal into a digital input signal, digital signal processing means for processing said digital input signal and generating a digital output signal, D/A conversion means for conversion of said digital output signal into an analog output signal and output signal transducer means for conversion of said analog output signal into an output sound signal perceivable to the user, a bidirectional communication link being provided between said units to connect a point in the signal path between the input signal transducer means and the digital signal processing means in one of said units with a corresponding point in the signal path between the input signal transducer means and the digital signal processing means of the other of said units, c h a r a c t e r i z e d in that the digital signal processing means of each unit (1, 2;16, 17) is arranged to effect a substantially full digital signal processing including individual processing of signals from the input transducer means (3r, 3l; 18r, 18l) of the actual unit and simulated processing of signals from the input transducer means of the other unit as well as binaural signal processing of signals supplied, on one hand, internally from the input signal transducer means of the same unit and, on the other hand, via said communication link (7; 28) from the input signal transducer means of the other unit, said digital signal processing means including at least a first digital signal processor part (5r, 5l, 12r, 12l;

21r-23r, 21l-23l) for processing said internally supplied signal, a second digital signal processor part (6l, 6r, 13l, 13r; 21ls-23ls, 21rs-23rs) for processing the signal supplied via said communication link (28) and a third digital signal processor part (9r, 9l; 24r-25r, 24l-25l) to effect common binaural digital signal processing of information derived from the signals processed in said first and second digital signal processor parts, said second digital signal processor part (6l, 6r, 13l, 13r; 21ls-23ls, 21rs-23rs) in each unit simulating the first digital signal processor part (5l, 5r, 12l, 12r; 21l-23l, 21r-23r) in the other unit with respect to adjustment parameters controlling the performance of said first signal processor part in said other unit.

2. A hearing aid system as claimed in claim 1, c h a r a c t e r i z e d in that said bidirectional communication link (7) is a wireless transmission link, transceiving means (8r, 8l) and antenna means (7r, 7l) being provided in each of said hearing aid units.

3. A hearing aid system as claimed in claim 2, c h a r a c t e r i z e d in that said antenna means (7r, 7l) in each hearing unit comprises a short piece of antenna wire projecting externally from an ear mold arranged in ITE or CIC position and acting simultaneously as a withdrawal string.

4. A hearing aid system as claimed in claim 1, 2 or 3, c h a r a c t e r i z e d in that in at least one of said hearing aid units (16, 17) adjustment parameters for processing units in an actual signal processing channel effecting signal processing adapted to the ear in which the unit is arranged as well as further adjustment parameters for the simulated signal processing channel effecting said simulated signal processing adapted to the ear in which the other unit

is arranged are entered into a memory (30).

5. A hearing aid system as claimed in claim 4, c h a r a c t e r i z e d in that said memory (30) further includes performance programme settings to provide varying operation modes ranging from fully binaural signal processing to simple monaural signal processing in the hearing aid units and/or provide adaption of system performance to different sound environments or listening situations.

6. A hearing aid system as claimed in claim 5, c h a r a c t e r i z e d in that at least one of said hearing aid units (1, 2; 16, 17) include means for calculation of intermediate settings between at least two consecutive performance programme settings.

7. A hearing aid system as claimed in claim 5 or 6, c h a r a c t e r i z e d in that programme selection means (31) for said performance programme settings including said intermediate settings are user operated and/or automatically activated on the basis of incoming sound signal analysis.

8. A hearing aid system as claimed in any of claims 2 or 3 and claim 7, c h a r a c t e r i z e d in that said selection means (31) are remotely controlled by wireless transmission of control signals.

9. A hearing aid system as claimed in any of claims 5 to 8, c h a r a c t e r i z e d in that said memory (30) is provided in one hearing aid unit only acting as a master unit, said bidirectional communication link (28) being designed for transfer of adjustment parameter settings from said one unit to the other unit acting as a slave unit.

10. A hearing aid system as claimed in any of claims 6, 7 or 8, c h a r a c t e r i z e d in that each of said hearing aid units (16, 17) is provided with said memory (30) and said programme selection

means (31), said selection means (31) being designed in both units for user operation as well as automatic activation, said bidirectional communication link (28) being designed for exchange of parameter settings between the two units and one of the units including decision means to provide for a consensus control of the system.

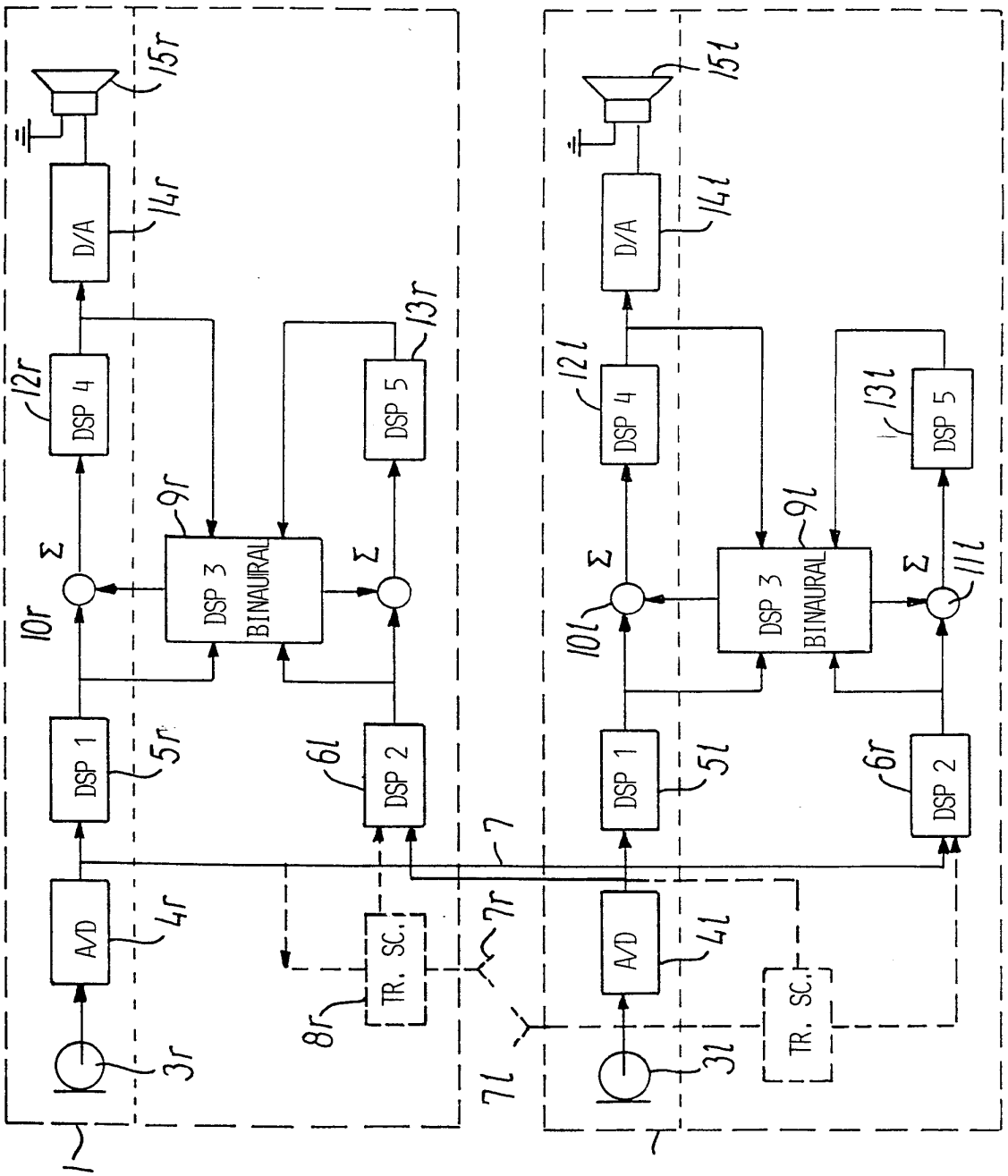
11. A hearing aid system as claimed in any of the preceding claims, c h a r a c t e r i z e d in that the binaural signal processing functions in each of the two hearing aid units are mirrored with respect to the binaural signal processing functions in the other unit to take account of sound level and/or sound spectrum differences between incoming sound signal at the right and left ear hearing aid units.

12. A hearing aid system as claimed in claim 11, c h a r a c t e r i z e d in that the binaural signal processing part of each of said hearing aid units comprises a feedback howl suppression system providing howl suppression by processing of a residual signal representing the difference between feedback signals in the actual and simulated signal processing channels.

13. A hearing aid system as claimed in any of any of the preceding claims, c h a r a c t e r i z e d in that in each of the hearing aid units means are provided for limitation, e.g. by compression of the signals supplied from said first and second processor parts to said third processor part.

14. A hearing aid system as claimed in any of the preceding claims, c h a r a c t e r i z e d in that means (33) are provided in each of the two hearing aid units for exchange of synchronization information between the signal processing parts of the two units.

FIG.1



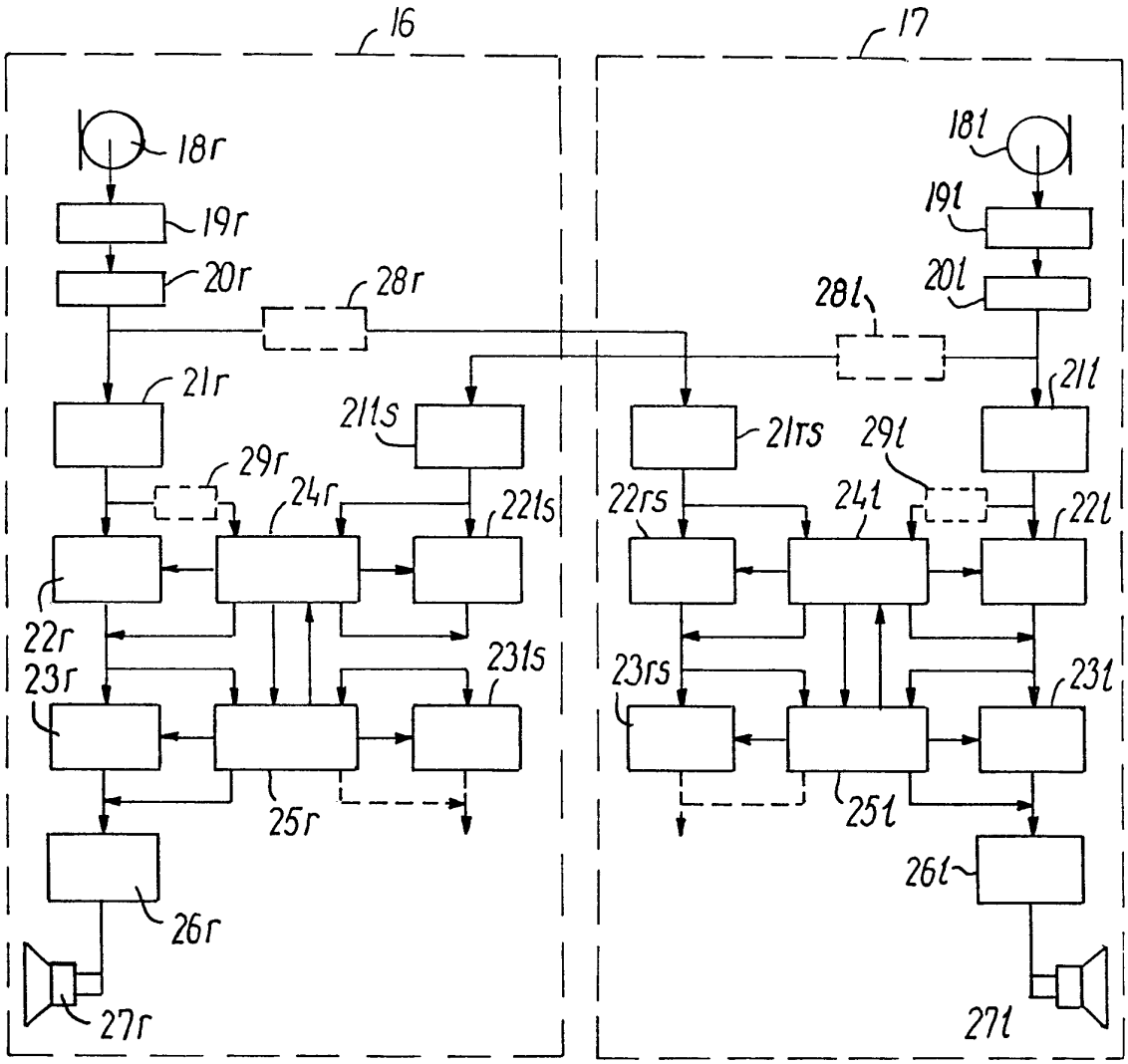

16
17
18r
19r
20r
28r
21r
211s
29r
24r
221s
22r
231s
23r
25r
26r
27r
181
191
201
28l
211
21rs
29l
22rs
241
221
23rs
231
251
261
271

FIG. 2

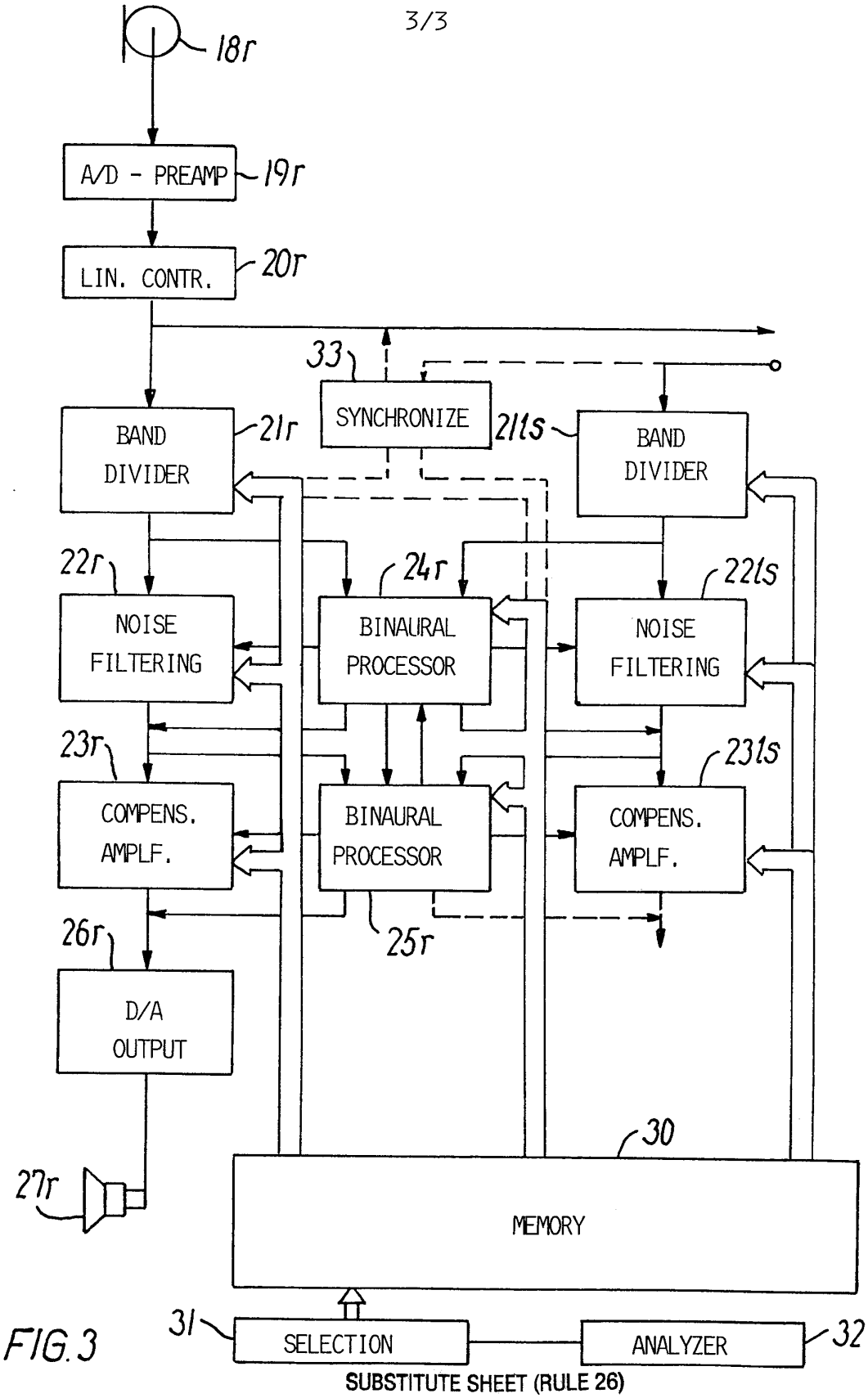

18r
A/D - PREAMP
19r
LIN. CONTR.
20r
33
SYNCHRONIZE
21r
BAND DIVIDER
21ls
BAND DIVIDER
22r
NOISE FILTERING
24r
BINAURAL PROCESSOR
22ls
NOISE FILTERING
23r
COMPENS. AMPLF.
BINAURAL PROCESSOR
25r
231s
COMPENS. AMPLF.
26r
D/A OUTPUT
27r
30
MEMORY
31
SELECTION
ANALYZER
32

FIG. 3

INTERNATIONAL SEARCH REPORT

In itional Application No
PCT/DK 98/00062

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04R25/00

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 6 H04R

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

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

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 97 14268 A (AUDIOLOGIC INC) 17 April 1997 cited in the application see page 3, line 5 - line 25 see page 4, line 29 - page 7, line 6; figures ---	1
A	US 5 511 128 A (LINDEMANN) 23 April 1996 see column 2, line 35 - column 4, line 19; figures ---	1
A	US 5 479 522 A (LINDEMANN ET AL.) 26 December 1995 cited in the application see column 2, line 11 - line 59 see column 3, line 50 - column 6, line 20 --- -/--	1

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

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

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

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

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

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 November 1998	11/11/1998

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Gastaldi, G

2

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INTERNATIONAL SEARCH REPORT

Int tional Application No
PCT/DK 98/00062

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	US 5 608 803 A (MAGOTRA ET AL.) 4 March 1997 see column 1, line 45 - column 2, line 46	1

2

Form PCT/ISA/210 (continuation of second sheet) (July 1992)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/DK 98/00062

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9714268 A	17-04-1997	US 5757932 A	26-05-1998
		AU 7248796 A	30-04-1997
		EP 0855130 A	29-07-1998
		NO 981559 A	08-06-1998
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		WO 9520305 A	27-07-1995
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US 5608803 A	04-03-1997	NONE	

Form PCT/ISA/210 (patent family annex) (July 1992)

Electronic Acknowledgement Receipt

EFS ID:	48591190
Application Number:	90019169
International Application Number:	
Confirmation Number:	7859
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	11244666
Customer Number:	176134
Filer:	Jacob Andrew Snodgrass
Filer Authorized By:	
Attorney Docket Number:	0027071.00133
Receipt Date:	15-SEP-2023
Filing Date:	24-FEB-2023
Time Stamp:	07:33:08
Application Type:	Reexam (Third Party)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Information Disclosure Statement (IDS) Form (SB08)	IDS_Form_1.pdf	1515313 3db40ffa2ea21abf52f1e0e58df31c9aac4d815b	no	20

Warnings:

Information:					
2	Non Patent Literature	2-3_Techniques_and_Applications_of_Wearable_Augmented_Reality_Audio.pdf	5851262 63e37914e8a53499dffeb9440d48687fb26103c4	no	20
Warnings:					
Information:					
3	Non Patent Literature	2-4_Interaction_Techniques_Using_Prosodic_Features_of_Speech_and_Audio_Localization.pdf	1673888 a8316a0ca6a97e58ff5cdefa0ae50524b1efca17	no	3
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4	Foreign Reference	2-10_WO2006-114101_Andersen.pdf	2110890 cbb5cbd667eae6cffbbb9feec0c16835f39ea537	no	54
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5	Non Patent Literature	2-18_A_Pattern_Recognition_Approach.pdf	309195 e1a6ba42e66713d81cecedc4451ed06fc2d6823d	no	12
Warnings:					
Information:					
6	Foreign Reference	2-25_EP1674061_Berg.pdf	173971 526cd3b19cee232f1d1ed856cfbb0427f793e8c4	no	12
Warnings:					
Information:					
7	Non Patent Literature	2-28_Audiometric_Ear_Canal_Probe_with_Active_Ambient_Noise_Control.pdf	1078748 9e6c4e7df57a09e94f0ad52487f1263b1465fefb	no	7
Warnings:					
Information:					
8	Foreign Reference	2-40_EP0643881_corresponding_document_WO9326084_Claybaugh.pdf	949658 b76ba1f333ad5fb8f4e7f01654b081ac012787ba	no	16
Warnings:					
Information:					

9	Foreign Reference	2-47_EP1640972_Dijkstra972.pdf	219543 c59a9f5c0541d826a9c97e8648599bcc869ba635	no	15
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10	Non Patent Literature	2-57_Learning_Based_Three_Dimensional_Sound.pdf	1052009 efad6bc1463d04b1d56094a6565426 8c66bdcce0	no	10
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11	Foreign Reference	2-58_EP1489596_Gustavsson596.pdf	97248 658c247f5acec77d5619b7ad75d933e1141a2e40	no	11
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12	Foreign Reference	2-62_JPH10294989_Hashimoto.pdf	3263303 62e83893e88f3e07092b50d37b3e9be7aece4984	no	41
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13	Foreign Reference	2-65_DE4312155_Hiller.pdf	677416 b52885bb20edf1e49781425ecbedc5d1bcfa0bb8	no	6
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14	Foreign Reference	2-66_WO93-26085_Hohman.pdf	1077990 1ae8b070fdf573f81bcd7ca4dc3ae5b4f0dea40a	no	19
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15	Foreign Reference	2-69_EP0935236_Housni.pdf	764899 554d61075e1a7e2f2d9ff4b7db58f6a2b5c5ae21	no	10
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16		90_019169_IDS_No_1_Cover_Letter_Corrected.pdf	705157 8062cc1040da73a4e39db023ee43c0299fa24dc5	yes	2
	Multipart Description/PDF files in .zip description				
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	Transmittal Letter		1	1	
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17	Foreign Reference	2-13_JP2005-227511_Arimune_Corrected.pdf	19593277 d3cb9eeac9487687f5e0fbae055934e42c57dcaf	no	22
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18	Non Patent Literature	2-77_Speech_Input_Hardware_Investigation_Corrected.pdf	10755602 ef402d273fec1a52df225aa7a420ba9d84547fd8	no	26
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19	Non Patent Literature	2-79_The_Effect_of_Hearing_Aid_Microphone_Location_Corrected.pdf	18152717 f2ca995fde59da9883422f8bb49950b38804864d	no	208
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20	Foreign Reference	2-9_JP2004-289762_Amada_Corrected_Reduced.pdf	10739708 d09446d7263cd4ec0aa4c23b3bdaa186105081b1	no	59
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21	Non Patent Literature	2-84_A_New_Two-sensor_Active_Noise_Cancellation_Algorithm.pdf	249096 def2413b8e8d0aaccc0cdce7d10aea2a2ea5a8e1	no	4
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22	Non Patent Literature	2-86_Optimal_Feedback_Control_Formulation_of_the_Active_Noise_Cancellation.pdf	7451828 65cd9155c24d2c616ae0fc000ecd1b3d4915dfe2	no	158
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23	Foreign Reference	2-87_JP297362_Katsumata.pdf	98573 d665247edb8872b39f0e98fbd494bf45cd190188	no	5
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24	Foreign Reference	2-91_KR20030058432_Kim432.pdf	461820 a8a65a5fe95c4c211c363affd6018832b4686acf	no	10
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25	Foreign Reference	2-92_KR20020086433_Kim433.pdf	851604 9769c9e8fac156ba5468d82d52bc2512c4af285f	no	12
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26	Foreign Reference	2-93_KR20030069471_Kim471.pdf	471231 f795a95a195a3d32449480361cb335e0a917cf8a	no	9
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27	Foreign Reference	2-94_JP3085237_Kondo237.pdf	386146 635f29c80e77d61cf7c5df73fa3c65956fe27750	no	7
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28	Foreign Reference	2-95_JP12878298_Kondo298.pdf	847811 5ee690c41b88b9ec1ecde8b8d4c5328930ef37ff	no	15
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29	Foreign Reference	2-96_JPH11331990_Kondo990.pdf	849864 88ef8349467cea8207e4e9fee807914f968819ea	no	15
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30	Non Patent Literature	2-99_Products_of_Interest.pdf	3665040 657eb1dba773009147496f3561ca779f1a2e6a8e	no	16
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31	Non Patent Literature	2-107_Spectral_Analysis_of_Speech_by_Linear_Prediction.pdf	1422522 f06b0aff1d55be78f897c3ee7324570c1824dd04	no	9
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32	Foreign Reference	2-111_JP3556987_Masui.pdf	440399 c7a6a555346a9460239ddd5212306bd037fd29e7	no	9
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33	Foreign Reference	2-115_WO2006-037156_Mejia156.pdf	2468560 2ee3372377294076db1fcbc85846685b35f74c2b	no	37
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34	Non Patent Literature	2-119_E-3_In-Flight_Acoustic_Exposure_Studies_and_Mitigation_via_Active_Noise_Reduction_Headset.pdf	3811407 f5941f58bbbf9c5f91ea4b3b6a2b8d7a0ac82fba	no	73
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35	Foreign Reference	2-121_JP2005168888_Murata.pdf	976208 79b02de01931bbd989dac1abde87e245d32b56a0	no	18
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36	Foreign Reference	2-122_WO01-57852_Nakadai.pdf	2537828 6048a06ed08312a95efdd1b48c6ce70067a79e2b	no	32
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37	Foreign Reference	2-126_AU2006-200446_Nemirovski446.pdf	3759384 08c862914e63cc3cc85017ff1558ae55c411d6a1	no	57
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38	Non Patent Literature	2-132_A_Binaural_Processor_for_Missing_Data.pdf	570132 e22b1a9933a4abaebe66ad40f84146451 3f66c07	no	18
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39	Non Patent Literature	2-110_Active_Noise_Cancellation_for_Headphones_Corrected.pdf	4546077 d1ae3676c1c1d52080064a7dac82ef88b7494595	no	74
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40	Non Patent Literature	2-118_Performance_of_Dual-Microphone_In-the-Ear_Hearing_Aids_Corrected.pdf	2690050 0711f2e4b1726e64c1d9171a2a4dc21626d9d856	no	9
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41	Non Patent Literature	2-129_In-Ear_Microphone_Techniques_for_Severe_Noise_Situations_Corrected.pdf	13167575 72770a0c3f87bab036ddac917a0d5f805c6f3af5	no	39
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42	Foreign Reference	2-135_EP1681903_Platz903.pdf	2014170 b88f88d6e73c538a6b78e89271bc616ae3f4178e	no	17
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43	Non Patent Literature	2-136_Adaptive_Noise_Cancellation_in_a_Multimicrophone_System.pdf	1462163 1af8e9a1c4323cfab9a48fce4ff537e2add0525c	no	11
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44	Non Patent Literature	2-137_Combined_feedback-feedforward_active_noise-reducing_headset.pdf	1877880 bfa6347aae29ddf345db7a8e8d26c8e89be98044	no	10
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45	Foreign Reference	2-144_EP1320281_Roeck.pdf	1661606 3ab6c82600bf9cce73ae8532ee91fe96b9cb6cde	no	20
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46	Foreign Reference	2-151_WO2006-054205_Schobben.pdf	1857259 68230bb2d64a6bd9e127ffe9ce5761c35106887a	no	24
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47	Non Patent Literature	2-152_Microphone_Array_Processing_for_Robust_Speech_Recognition.pdf	1077968 c41e9a7f76dd756f67eeca6fae407c484fb2d1f5	no	163
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48	Foreign Reference	2-154_EP1638079_Sheikhzadeh.pdf	2944969 5f400ef89c2c8e441adccd41fd5656faf07c19ca	no	49
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49	Foreign Reference	2-155_WO86-00133_Spalding.pdf	1629242 654ed76e98b996d01bd146229e5d9a5f19061756	no	26
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50	Foreign Reference	2-158_EP1594344_Stirnemann.pdf	1243541 8ade69b724f7317c3e78a6e83c530e0e44f74555	no	16
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51	Foreign Reference	2-163_WO02-17836_Svean836.pdf	2920659 07f73abcb6fc80149c3ce290c19e8beec21c917d	no	41
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Information:					
52	Non Patent Literature	2-164_Sound_Source_Localization_and_Separation.pdf	678723 5dca2301ee2a892336e0fbc9386bccb40d37b123	no	53
Warnings:					
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53	Foreign Reference	2-170_EP1519625_Victorian625.pdf	1676288 0c2fa393d9bfd62db3533d38bb8cfae6b913628d	no	21
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54	Non Patent Literature	2-140_A_Dual-Mode_Human-Machine_Interface_for_Robotic_Control_Corrected.pdf	8729924 02303f33063037325371b1b8acd826faf9ff22bce	no	7
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55	Non Patent Literature	2-143_Isolated_Word_Recognition_from_In-Ear_Microphone_Data_Corrected_Reduced.pdf	24011764 60b2d9f09c3a956918c5247d48bb46c1bfb12afc	no	179
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56	Non Patent Literature	2-147_Active_Noise_Control_Low-Frequency_Techniques_Corrected.pdf	5165743 b3d13358be5dd26fb40d2b5d3ddf7d641f68e419	no	24
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57	Non Patent Literature	2-148_Reducing_the_Negative_Effects_of_Ear-Canal_Occlusion_Corrected.pdf	9772618 3b159f1aae4a5e63d9d86003af496deec61a28c2	no	6
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58	Non Patent Literature	2-159_Multi-Microphone_Correlation-Based_ProcessingCorrected_Reduced.pdf	18207429 9b643d632d7eae3a1bd83ee1d33006ad03fb5c6e	no	113
Warnings:					
Information:					
59	Foreign Reference	2-176_WO99-43185_Westermann.pdf	2288093 b151500334e410117ede6a671fcc3a3996e14388	no	31
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(Not for submission under 37 CFR 1.99)

Application Number	90019169
Filing Date	2023-02-24
First Named Inventor	Goldstein et al.
Art Unit	3992
Examiner Name	Ge, Yuzhen
Attorney Docket Number	3-00051-666EPR

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	1	NACRE QuietPro User Manual v2.0	
	2	STRAND, et al. On the Feasibility of ASR in Extreme Noise Using the Parat Earplug Communication Terminal, IEEE, 2003, pp. 315-320	
	3	New Scientist Article_Tuning in, February 14, 2001, (https://www.newscientist.com/article/dn423-tuning-in/)	
	4	MUGGLETON, Nacre's QUIETPRO+ Intelligent Hearing System: Delivering Performance and Hearing Protection Now for the Future of the Soldier, SoldierMod Jan. 2009, pp. 28-29, Vol. 2	
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	9	Nacre v Silynx, TTAB Notice of Opposition	
	10	Delivery Order W912DQ06D0037-0001	
	11	Nacre v Silynx, TTAB Applicant's Answer	

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	14	Jawbone Aliph User Manual	
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	16	CNET Article - Aliph Jawbone Bluetooth headset review, Jan. 31, 2007 (https://www.cnet.com/reviews/aliph-jawbone-bluetooth-headset-review/)	
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	21	Gennum Zen User Manual	
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	34	Plantronics MX200 User Guide	
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	Attorney Docket Number	3-00051-666EPR

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7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Electronic Acknowledgement Receipt

EFS ID:	48591216
Application Number:	90019169
International Application Number:	
Confirmation Number:	7859
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	11244666
Customer Number:	176134
Filer:	Jacob Andrew Snodgrass
Filer Authorized By:	
Attorney Docket Number:	0027071.00133
Receipt Date:	15-SEP-2023
Filing Date:	24-FEB-2023
Time Stamp:	07:48:16
Application Type:	Reexam (Third Party)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		90_019169_IDS_No_2_Cover_Letter_Corrected.pdf	2244728 21ac45a1ecc3a50f5540d60c08527dfb974b6e70	yes	5

	Multipart Description/PDF files in .zip description				
	Document Description	**Start**	**End**		
	Transmittal Letter	1	4		
	Reexam Certificate of Service	5	5		
Warnings:					
Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	IDS_Form_2.pdf	1338538 0ab8f48ed4d4fac9546872c605069b6f718e978f	no	8
Warnings:					
Information:					
A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
3	Non Patent Literature	3-1_SAM-TECH_00055181_QuietPro_Manual_v_2_0.pdf	12126834 9424666fb165a29fe816a473bb547dafc5d3c48a	no	48
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4	Non Patent Literature	3-2_SAM-TECH_00063687_IEEE_Explore_Article.pdf	638136 e41bbbf3e551dd41f44797bbb674fc98037e82a0	no	6
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5	Non Patent Literature	3-3_SAM-TECH_00064068_New_Scientist_Article.pdf	1131223 fdb4f9f56a16d54ec4799980d3772265da2e4450	no	2
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6	Non Patent Literature	3-4_SAM-TECH_00065729_SoldierMod_Article.pdf	1515436 5bf1bb6540e3177d181624cff63416a5487e9193	no	2
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7	Non Patent Literature	3-5_SAM-TECH_00097600_Quietpro_protects_and_enhances_hearing.pdf	2158443 2d8c079943d00140f398e903e345a3dde9728872	no	6
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8	Non Patent Literature	3-6_SAM-TECH_00063985_Honeywell_Quietpro_QP100ex.pdf	16701328 2a5f92f127fd40dda021310ee117554d4aa4e784	no	33
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9	Non Patent Literature	3-7_SAM-TECH_00054131_Nacre_WBM.pdf	1696652 7a3c0b00e543e10c0b1702a46ab62b6b354653f2	no	2
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10	Non Patent Literature	3-8_SAM-TECH_00054652_Quietpro_QP100ex.pdf	16711549 aca866a19adb7c67e57b7f0c6fd7a33a6c636024	no	33
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11	Non Patent Literature	3-9_SAM-TECH_00054696_Nacre_v_Silynx_Notice_of_Opposition.pdf	1314759 7051b3ce794abcbdb6f3f1b834f91a290ded667	no	9
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12	Non Patent Literature	3-10_SAM-TECH_00055735_Delivery_Order.pdf	909577 5b20a77983ba6d96e318077fd01ee0fbac13e9ee	no	3
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13	Non Patent Literature	3-11_SAM-TECH_00052371_Nacre_v_Silynx_Apps_Answer.pdf	596552 a17a916f4c3ca2c5611bcaf2c79afbd7d9dd84da	no	6
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14	Non Patent Literature	3-12_SAM-TECH_00057150_Silynx_integrates_GPS.pdf	867626 167374cc011cc695e580409c9ecab50cc1a297ff	no	2
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15	Non Patent Literature	3-13_SAM-TECH_00062036_Military_Headset_Reaches_Masses.pdf	2377294 e0a27344f1d666f58ac1bc245180cdc8dddfb8c5	no	7
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16	Non Patent Literature	3-14_SAM-TECH_00061992_Jawbone_User_Manual.pdf	1075935 dc9996c525be9d17f0fc2360614d0c6c8fba5d57	no	16
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17	Non Patent Literature	3-15_SAM-TECH_00062054_New_earphones_let_you_go.pdf	465620 a95528e62eea9b01d9c6d28aa406625821abb2ea	no	2
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18	Non Patent Literature	3-16_SAM-TECH_00060121_Aliph_Jawbone_Bluetooth_headset_review.pdf	2005419 e974ec681f14f16ff4b3697cee33d5c4a9240b95	no	3
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19	Non Patent Literature	3-17_SAM-TECH_00094798_beoworld_Earset_2_Bluetooth_Headset.pdf	1316015 d5de33f79619d9d899a25e29c1529b55b724c775	no	3
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20	Non Patent Literature	3-18_SAM-TECH_00094865_dexigner_news_Earset_2.pdf	2691755 1ee2be35ae5749aca6c8cd4ea0d8fd579ecc623b	no	7
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21	Non Patent Literature	3-19_SAM-TECH_00095178_ER6i_TS.pdf	586640 23ed2f68dd4c2124921b1dd5b6dad88780f73d8c	no	1
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22	Non Patent Literature	3-20_SAM-TECH_00095121_Etymotic_ER-6_review.pdf	3152047 0b7be0bdcc256ac39794589ffcbecec46f9f58fc	no	5
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23	Non Patent Literature	3-21_SAM-TECH_00098432_Gennum_Zen_User_Manual.pdf	4211264 282fdef6c1d9363d2d73dbe5fa30384a86df47ba	no	44
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24	Non Patent Literature	3-22_SAM-TECH_00098419_CNET_Article_Gennum_Zen_Bluetooth_Headset_Review.pdf	2264679 c808c9cdc1659d71cca13937f5b737809cfa11aa	no	4
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25	Non Patent Literature	3-23_SAM-TECH_00098485_Globe_and_Mail_Article_Gennum_Zen_Headset_for_Bluetooth.pdf	3401890 86dfac16ea7780bfe03271a9b32a3f434ae70ff1	no	5
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Information:					
26	Non Patent Literature	3-24_SAM-TECH_00098490_PocketLint_Article-Zen_Gennum_Bluetooth_Headset.pdf	5669287 fad91de375ee9c3892a35bae14d1e0d3c1c26ab2	no	13
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27	Non Patent Literature	3-25_SAM-TECH_00098719_Motorola_H605_User_Manual.pdf	3919945 e4017947e32680a5c5ba8ce75319aa9b6fab45ed	no	5
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28	Non Patent Literature	3-26_SAM-TECH_00098639_CNET_Article-Motorola_H605_Bluetooth_Headset_Review.pdf	1666588 72b0132f5427cf64663a98098818e3ba06acc9de	no	4
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29	Non Patent Literature	3-27_SAM-TECH_00098743_PhoneArena_Article-Motorola_H605_Review.pdf	3344847 20cbdd4734afc67d0dc545308fa47cf2392aa7d3	no	12
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30	Non Patent Literature	3-28_SAM-TECH_00099254_Peltor_Lite-Com_II_Manual.pdf	1580888 89f01c687a160e562ea2b5a42fdbcee6e1c9f92d	no	13
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31	Non Patent Literature	3-29_SAM-TECH_00099203_Peltor_Lite-Com_II_Brochure.pdf	4504675 fafed3d728e570229fd5d20f1563ea991b329618	no	6
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32	Non Patent Literature	3-30_SAM-TECH_00099344_Plantronics_Discovery_655_User_Guide.pdf	4454702 55947d9603b41733f724d2f3df9ccb4f283e8a6c	no	16
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33	Non Patent Literature	3-31_SAM-TECH_00099296_Plantronics_Discovery_655_Brochure.pdf	1738148 5f296ed2809b11b5855aa78074166d51959d776c	no	2
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34	Non Patent Literature	3-32_SAM-TECH_00099287_CNET_Article-Plantronics_Discovery_655_Bluetooth_Headset_Review.pdf	1989832 722f1dc03db1e8f8dee822c5cfe983c3332a8481	no	4
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35	Non Patent Literature	3-33_SAM-TECH_00099387_Silicon_Poip_Culture_Article-Plantronics_Discovery_655.pdf	3319821 4a29a76a3c1705c8721e6b8c73dc281477520750	no	4
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36	Non Patent Literature	3-34_SAM-TECH_00099435_Plantronics_MX200_User_Guide.pdf	978419 4df0064095211a3b7c9a550decbf4463eec1a4f8	no	2
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37	Non Patent Literature	3-35_SAM-TECH_00099461_Plantronics_MX250_User_Guide.pdf	883602 c60a272ed449e59239311e81f6419e2053e98397	no	2
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38	Non Patent Literature	3-36_SAM-TECH_00099419_Plantronics_MX200_Brochure.pdf	1413096 0bf062e4bff9311598bad1aa3942b388565b9353	no	2
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39	Non Patent Literature	3-37_SAM-TECH_00099557_Sony_Walkman_User_Manual.pdf	15676010 91b6417a044cfd5a272bf212340845a26369a616	no	22
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40	Non Patent Literature	3-38_SAM-TECH_00099514_EAFIT_Article-The_Sony_Walkman.pdf	2270583 be8990ff9638112e59e300371990d71a50e92a73	no	9
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41	Non Patent Literature	3-39_SAM-TECH_00099533_IDG_Article-Sony_New_Walkman_Players_Pack_Noise_Canceling.pdf	1157259 a4ff902ef6c6eab4085317539b4eac57cb620171	no	3
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42	Non Patent Literature	3-40_SAM-TECH_00099579_Stuff_Article-Sony_NW-S700_Review.pdf	2266418 27657b737ed801fe394b2b97c85899318a5dacbe	no	8
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43	Non Patent Literature	3-41_SAM-TECH_00060368_CES_2006-Bluetooth_innovations_abound.pdf	5511586 2c41b9556d291c2cc8979ab311379e5ed7b4947e	no	9
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44	Non Patent Literature	3-42_SAM-TECH_00060509_Motorola_H9_Bluetooth_Headset_User_Manual.pdf	5030887 400e55001ce1f28c92bf64e3db2f1be861a16794	no	6
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45	Non Patent Literature	3-43_SAM-TECH_00056060_Motorola_Miniblue_Press_Release.pdf	951802 61b11660b33b02d7e5fdfb146f5900a16218317a	no	1
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46	Non Patent Literature	3-44_SAM-TECH_00060628_Engadget_Article.pdf	1224208 622da4af408cfba510c71276c828acdb17fe5b24	no	3
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47	Non Patent Literature	3-45_SAM-TECH_00060424_CNET_Article-Motorola_H5_Miniblue_Blueotooth_Headset.pdf	2617551 4eb0ca1757bee0ebe22eabb9b8c8ede85bbc1417	no	5
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48	Non Patent Literature	3-46_SAM-TECH_00100046_Sensaphonics_3D_Active_Ambient_IEM_System_User_Guide.pdf	4846524 02502175d7c0b2070959d52986bfa4f2f772968f	no	16
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49	Non Patent Literature	3-47_SAM-TECH_00100065_Sensaphonics_3D_Active_Ambient_IEM_System_Article.pdf	1350083 8b8a5356a72c92ece947c8c1983ad31503c3aee5	no	1
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50	Non Patent Literature	3-48_SAM-TECH_00096723_Products_of_Interest_Article.pdf	3665040 6ad19b3310ffcdeebc2896837f3ba9e45e9a4ea4	no	16
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Information:					
		Total Files Size (in bytes):	165531740		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

<u>New Applications Under 35 U.S.C. 111</u>
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.
<u>National Stage of an International Application under 35 U.S.C. 371</u>
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.
<u>New International Application Filed with the USPTO as a Receiving Office</u>
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Control No.	90/019,169
U.S. Patent No.	11,244,666
Reexamination Filing Date	February 24, 2023
Inventors	Goldstein et al.
Title	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
Examiner	Ge, Yuzhen
Art Unit	3992
Confirmation No.	7859

Submitted via EFS-Web
Mail Stop Ex Parte Reexamination
Attn: Central Reexamination Unit
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

TRANSMITTAL LETTER FOR INFORMATION DISCLOSURE STATEMENT

Patent Owner herewith submits an Information Disclosure Statement (form PTO/SB/08a). For the convenience of the Examiner, Patent Owner provides the following table correlating the Non-Patent Literature Documents submitted herewith with the list of such references in the Information Disclosure Statement by Bates number, where applicable.

Cite No.	Identification	Bates No(s).
1	NACRE QuietPro User Manual v2.0	SAM-TECH_00055181 - SAM-TECH_00055228
2	STRAND, et al. On the Feasibility of ASR in Extreme Noise Using the Parat Earplug Communication Terminal, IEEE, 2003, pp. 315-320	SAM-TECH_00063687 - SAM-TECH_00063692
3	New Scientist Article_Tuning in, February 14, 2001, (https://www.newscientist.com/article/dn423-tuning-in/)	SAM-TECH_00064068 - SAM-TECH_00064069
4	MUGGLETON, Nacre's QUIETPRO+ Intelligent Hearing System: Delivering Performance and Hearing Protection Now for the Future of the Soldier, SoldierMod Jan. 2009, pp. 28-29, Vol. 2	SAM-TECH_00065729 - SAM-TECH_00065730
5	TUNMO, QuietPro Protects and enhances hearing, Oct. 23, 2006 (available at https://www.tu.no/artikler/quietproverner-og-forsterker-horselen/261960)	SAM-TECH_00097600 - SAM-TECH_00097605

Cite No.	Identification	Bates No(s).
6	Honeywell QuietPro QP100ex Presentation, March 7, 2013 (available at https://studylib.net/doc/5501651/quietpro-qp100ex)	SAM-TECH_00063985 - SAM-TECH_00064017
7	Nacre website (htts://web.archive.org/web/20050309073333/http:/www.nacre.no:80/)	SAM-TECH_00054131 - SAM-TECH_00054132
8	Honeywell QuietPro QP100ex Presentation, March 7, 2013 (available at https://studylib.net/doc/5501651/quietpro-qp100ex)	SAM-TECH_00054652 - SAM-TECH_00054684
9	Nacre v Silynx, TTAB Notice of Opposition	SAM-TECH_00054696 - SAM-TECH_00054704
10	Delivery Order W912DQ06D0037-0001	SAM-TECH_00055735 - SAM-TECH_00055737
11	Nacre v Silynx, TTAB Applicant's Answer	SAM-TECH_00052371 - SAM-TECH_00052376
12	Silynx Integrates GPS in Headset for Special Operations, May 13, 2008, (https://defense-update.com/20080513_c4ops.html)	SAM-TECH_00057150 - SAM-TECH_00057151
13	Military Headset Reaches Masses, Sep. 2004 (available at https://www.wired.com/2004/09/military-headset-reaches-masses)	SAM-TECH_00062036 - SAM-TECH_00062042
14	Jawbone Aliph User Manual	SAM-TECH_00061992 - SAM-TECH_00062007
15	New earphones let you go cordless on iPods, cell phones, Dec. 24, 2006, (https://www.capecodtimes.com/story/news/2006/12/24/new-earphones-let-you-go/50845129007/)	SAM-TECH_00062054 - SAM-TECH_00062055
16	CNET Article - Aliph Jawbone Bluetooth headset review, Jan. 31, 2007 (https://www.cnet.com/reviews/aliph-jawbone-bluetooth-headset-review/)	SAM-TECH_00060121 - SAM-TECH_00060123
17	EarSet 2 Bluetooth Headset (available at https://www.beoworld.org/prod_details.asp?pid=733)	SAM-TECH_00094798 - SAM-TECH_00094800
18	EarSet 2: Tonal Precision and Superb Comfort From Bang & Olufsen, Jan. 2, 2007, (https://www.dexigner.com/news/9935)	SAM-TECH_00094865 - SAM-TECH_00094871
19	Etymotic ER-6i Technical Specifications (available at https://www.etymotic.com/ephp/er6i-ts.aspx)	SAM-TECH_00095178
20	CNET Article-Etymotic ER-6 review, Feb. 7, 2005, (https://www.cnet.com/reviews/etymotic-er-6-review/)	SAM-TECH_00095121 - SAM-TECH_00095125
21	Gennum Zen User Manual	SAM-TECH_00098432 - SAM-TECH_00098475
22	CNET Article, Gennum Zen Bluetooth Headset Review, July 1, 2005	SAM-TECH_00098419 - SAM-TECH_00098422

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23	Globe and Mail Article - Gennum Z-E-N Headset for Bluetooth, Dec. 7, 2004	SAM-TECH_00098485 - SAM-TECH_00098489
24	PocketLint Article - Zen Gennum Bluetooth Headset, February 3, 2005	SAM-TECH_00098490 - SAM-TECH_00098502
25	Motorola H605 User Manual	SAM-TECH_00098719 - SAM-TECH_00098723
26	CNET Article - Motorola H605 Bluetooth Headset Review, July 18, 2006	SAM-TECH_00098639 - SAM-TECH_00098642
27	PhoneArena Article - Motorola H605 Review, Sep. 9, 2008	SAM-TECH_00098743 - SAM-TECH_00098754
28	Peltor LiteCom Pro II Headset Manual	SAM-TECH_00099254 - SAM-TECH_00099266
29	Peltor Lite-Com II Brochure	SAM-TECH_00099203 - SAM-TECH_00099208
30	Plantronics Discovery 655 User Guide	SAM-TECH_00099344 - SAM-TECH_00099359
31	Plantronics Discovery 655 Brochure	SAM-TECH_00099296 - SAM-TECH_00099297
32	CNET Article - Plantronics Discovery 655 Bluetooth Headset Review, Aug. 23, 2006	SAM-TECH_00099287 - SAM-TECH_00099290
33	Silicon Poip Culture Article - Plantronics Discovery 655, May 28, 2007	SAM-TECH_00099387 - SAM-TECH_00099390
34	Plantronics MX200 Brochure	SAM-TECH_00099435 - SAM-TECH_00099436
35	Plantronics MX250 User Guide	SAM-TECH_00099461 - SAM-TECH_00099462
36	Plantronics MX200 Brochure	SAM-TECH_00099419 - SAM-TECH_00099420
37	Sony Walkman User Manual	SAM-TECH_00099557 - SAM-TECH_00099578
38	EAFIT Article - The Sony Walkman	SAM-TECH_00099514 - SAM-TECH_00099522
39	IDG Article - Sony's New Walkman Players Pack Noise Canceling, Oct. 13, 2006	SAM-TECH_00099533 - SAM-TECH_00099535
40	Stuff Article - Sony NW-S700 Review, Jan. 17, 2007, (https://www.stuff.tv/review/sony-nw-s700-review/)	SAM-TECH_00099579 - SAM-TECH_00099586
41	New Atlas Article - CES 2006: Bluetooth innovations abound - Inner-ear headset, bluetooth keyboard and wireless iPod companion, (https://www. newatlas.com /ces-2006-bluetooth-innovations-abound -inner-ear-headset- bluetooth-keyboard-and-wireless-ipod-companion/4977/)	SAM-TECH_00060368 - SAM-TECH_00060376
42	Motorola H9 Bluetooth Headset User Manual	SAM-TECH_00060509 - SAM-TECH_00060514

Cite No.	Identification	Bates No(s).
43	Motorola Miniblue Press Release (https://web.archive.org/web/20060212115000/http://www.motorola.com/motoinfo/product/details/0,,133,00.html)	SAM-TECH_00056060
44	Engadget Article - Motorola's H5 Miniblue Bluetooth Headset, Jan. 2006, (https://www.engadget.com/2006-01-04-motorolas-h5-miniblue-bluetooth headset. html)	SAM-TECH_00060628 - SAM-TECH_00060630
45	CNET Article - Motorola H5 Miniblue Bluetooth Headset, Jan. 4, 2006, (https://www.cnet.com/tech/mobile/motorolah5-miniblue-bluetooth-headset/)	SAM-TECH_00060424 - SAM-TECH_00060428
46	Sensaphonics 3D Active Ambient IEM (In-Ear Monitor) System User Guide	SAM-TECH_00100046 - SAM-TECH_00100061
47	Sensaphonics 3D Active Ambient IEM System Article, Feb. 2006 (https://www.mixonline.com/news/sensaphonics-3d-active- ambient-iem-system-421859)	SAM-TECH_00100065
48	Kyle Gann, Otto E. Laske, Colby Leider, Products of Interest, Computer Music Journal, Vol. 30, Number 3, Fall 2006, pp. 93-107 (Article)	SAM-TECH_00096723 - SAM-TECH_00096738

Patent owner identifies the references submitted herewith by certain indicia that appear on the references, for example, dates. Patent Owner does not warrant that such references were published or otherwise publicly available on such dates or that any of the identified references constitutes prior art to the '666 patent.

Patent Owner submits herewith the IDS fee. Should any additional fees be due in connection with this paper or at any other time during this proceeding, the Commissioner is authorized to charge such fees to Deposit Account No. 60-3815.

Dated: September 15, 2023

Respectfully submitted,

/s/ Jacob A. Snodgrass
Jacob A. Snodgrass
Registration No. 66,032

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CERTIFICATE OF SERVICE

Pursuant to 37 C.F.R. § 1.550(f), the undersigned hereby certifies that on September 15, 2023, a copy of this letter as well as the Information Disclosure Statement (form PTO/SB/08a) and cited non-patent literature documents (on flash drive) were served on Third Party Requester at the following address.

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Dated: September 15, 2023

Respectfully submitted,

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PV LAW LLP
Customer No. 176134
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Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

PTO/SB/08a (01-22)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**		
	Application Number	90019169
	Filing Date	2023-02-24
	First Named Inventor	Goldstein et al.
	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

U.S.PATENTS Remove

Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
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Examiner Initial*	Cite No	Publication Number	Kind Code[1]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
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Examiner Initial*	Cite No	Foreign Document Number[3]	Country Code[2] i	Kind Code[4]	Publication Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear	T[5]
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Examiner Initials*	Cite No	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc), date, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]

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	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	1	In-Ear Microphone Techniques for Severe Noise Situations Claim Chart (Exhibit N50 to Samsung's Invalidity Contentions)	
	2	Multi-Sensory Microphones for Robust Speech Detection, Enhancement and Recognition Claim Chart (Exhibit N51 to Samsung's Invalidity Contentions)	
	3	Opening Expert Report of Nathaniel Polish, Ph.D. Regarding Invalidity of U.S. Patent Nos. 8,315,400; 9,491,542; 9,609,424; 10,979,836; and 11,244,666, Volume I ('542 and '666 Patents) (redacted, excerpts re '666 patent) Feb. 6, 2023	
	4	Samsung's Motion for Summary Judgment of Invalidity under 35 U.S.C. §112 of the '666 Patent (Redacted) (Dkt. No. 457)	
	5	Techiya's Opposition to Samsung's Motion for Summary Judgment of Invalidity under 35 U.S.C. § 112 of the '666 Patent (Redacted) (Dkt. No. 515)	
	6	Samsung's Reply in Support of Its Motion for Summary Judgment of Invalidity under 35 U.S.C. § 112 of the '666 Patent (Redacted) (Dkt. No. 556)	
	7	Techiya's Sur-Reply to Samsung's Motion for Summary Judgment of Invalidity under 35 U.S.C. § 112 of the '666 Patent (Dkt. No. 587)	

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Examiner Signature		Date Considered	

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through a citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

☒ The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Jacob A. Snodgrass/	Date (YYYY-MM-DD)	2023-09-15
Name/Print	Jacob A. Snodgrass	Registration Number	66,032

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

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The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Electronic Acknowledgement Receipt

EFS ID:	48591398
Application Number:	90019169
International Application Number:	
Confirmation Number:	7859
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	11244666
Customer Number:	176134
Filer:	Jacob Andrew Snodgrass
Filer Authorized By:	
Attorney Docket Number:	0027071.00133
Receipt Date:	15-SEP-2023
Filing Date:	24-FEB-2023
Time Stamp:	08:59:41
Application Type:	Reexam (Third Party)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		90_019169_IDS_No_4_Cover_Letter_Corrected.pdf	694522 5d75c1941f5badd7a7a393c0198dd2b1a98b02cf	yes	2

	Multipart Description/PDF files in .zip description				
	Document Description	**Start**	**End**		
	Transmittal Letter	1	1		
	Reexam Certificate of Service	2	2		
Warnings:					
Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	IDS_Form_4.pdf	1347782 b9a7d3eb20aaff3c2517d8e160e16bff289ecf39	no	4
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A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
3	Non Patent Literature	1_Exhibit-N50_Westerlund_Corrected.pdf	1133079 10af352dcd5cfec01109a8b2b65682dc78830616	no	154
Warnings:					
Information:					
4	Non Patent Literature	2_Exhibit-N51_Zhang_Corrected.pdf	867605 feff8ab4cd929b401a67ba2685e32a4b33d23054	no	138
Warnings:					
Information:					
5	Non Patent Literature	3_Opening_Expert_Report_of_Nathaniel_Polish_Vol_1_Redacted_excerpts.pdf	6703209 52bd6dcbfac0a2f5d6c59ec9a6ccafe6e5e1231c	no	214
Warnings:					
Information:					
6	Non Patent Literature	4_Samsung_Motion_for_SJ_of_Invalidity_under_112_Redacted.pdf	1081493 50e9b138d08ab5de4e15d2bc9593560c6be05659	no	13
Warnings:					
Information:					

7	Non Patent Literature	5_Techiya_Opposition_to_Samsung_Motion_for_SJ_of_Invalidity_under_112_Redacted.pdf	693348 416a4559e14c27daf0f6ae315cce384f659aa11e	no	11
Warnings:					
Information:					
8	Non Patent Literature	7_Techiya_Sur-Reply_to_Samsung_Motion_for_SJ_of_Invalidity_under_112_Redacted.pdf	176580 e53919014c85ac307dd6d66ef2ce9d3b40060cf8	no	5
Warnings:					
Information:					
9	Non Patent Literature	6_Samsung_Reply_in_Support_of_Its_Motion_for_SJ_of_Invalidity_under_112_Redacted_Corrected.pdf	130114 880bf59d62490311fe30d1219a31850d5aacd135	no	7
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New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Control No.	90/019,169
U.S. Patent No.	11,244,666
Reexamination Filing Date	February 24, 2023
Inventors	Goldstein et al.
Title	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
Examiner	Ge, Yuzhen
Art Unit	3992
Confirmation No.	7859

Submitted via EFS-Web
Mail Stop Ex Parte Reexamination
Attn: Central Reexamination Unit
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

TRANSMITTAL LETTER FOR INFORMATION DISCLOSURE STATEMENT

Patent Owner herewith submits an Information Disclosure Statement (form PTO/SB/08a).

Patent Owner submits herewith the IDS fee. Should any additional fees be due in connection with this paper or at any other time during this proceeding, the Commissioner is authorized to charge such fees to Deposit Account No. 60-3815.

Dated: September 15, 2023

Respectfully submitted,

/s/ Jacob A. Snodgrass
Jacob A. Snodgrass
Registration No. 66,032

PV LAW LLP
Customer No. 176134
5335 Wisconsin Avenue, N.W., Suite 440
Washington, D.C. 20015
T: (202) 371-6861
F: (202) 888-3163

CERTIFICATE OF SERVICE

Pursuant to 37 C.F.R. § 1.550(f), the undersigned hereby certifies that on September 15, 2023, a copy of this letter as well as the Information Disclosure Statement (form PTO/SB/08a) and cited non-patent literature documents (on flash drive) were served on Third Party Requester at the following address.

Ali R. Sharifahmadian
Arnold & Porter Kaye Scholer LLP
601 Massachusetts Ave., N.W.
Washington, DC 20001

Dated: September 15, 2023

Respectfully submitted,

/s/ Jacob A. Snodgrass
Jacob A. Snodgrass
Registration No. 66,032

PV LAW LLP
Customer No. 176134
5335 Wisconsin Avenue, N.W., Suite 440
Washington, D.C. 20015
T: (202) 371-6861
F: (202) 888-3163

Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

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(Not for submission under 37 CFR 1.99)

Application Number	90019169
Filing Date	2023-02-24
First Named Inventor	Goldstein et al.
Art Unit	3992
Examiner Name	Ge, Yuzhen
Attorney Docket Number	3-00051-666EPR

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Examiner Initial*	Cite No	Patent Number	Kind Code[1]	Issue Date	Name of Patentee or Applicant of cited Document	Pages,Columns,Lines where Relevant Passages or Relevant Figures Appear
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Examiner Initials*	Cite No	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc), date, pages(s), volume-issue number(s), publisher, city and/or country where published.	T[5]

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	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	1	Samsung's Invalidity Contentions and P.R. 3-3 and 3-4 Disclosures for U.S. Patent Nos. 11,039,259, 11,057,701, 11,217,237, and 11,244,666 (Case No. 2:22-cv-00053-JRG-RSP) (excerpts re '666 patent) July 6, 2022	
	2	U.S. Pub. No. 2010/0119077 to Platz Claim Chart (Exhibit N1 to Samsung's Invalidity Contentions)	
	3	U.S. Patent No. 6,728,385 to Kvaløy Claim Chart (Exhibit N2 to Samsung's Invalidity Contentions)	
	4	U.S. Patent No. 5,276,740 to Inanaga Claim Chart (Exhibit N3 to Samsung's Invalidity Contentions)	
	5	U.S. Pub. No. 2007/0189544 to Rosenberg Claim Chart (Exhibit N4 to Samsung's Invalidity Contentions)	
	6	U.S. Pub. No. 2007/0021958 to Visser Claim Chart (Exhibit N5 to Samsung's Invalidity Contentions)	
	7	U.S. Pub. No. 2008/0165988 to Terlizzi Claim Chart (Exhibit N6 to Samsung's Invalidity Contentions)	
	8	U.S. Pub. No. 2003/0035551 to Light Claim Chart (Exhibit N7 to Samsung's Invalidity Contentions)	
	9	U.S. Pub. No. 2007/0223717 to Boersma Claim Chart (Exhibit N8 to Samsung's Invalidity Contentions)	
	10	U.S. Pub. No. 2007/0100637 to McCune Claim Chart (Exhibit N9 to Samsung's Invalidity Contentions)	
	11	U.S. Pub. No. 2007/0253569 to Bose Claim Chart (Exhibit N10 to Samsung's Invalidity Contentions)	

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	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	12	U.S. Pub. No. 2006/0083387 to Emoto Claim Chart (Exhibit N11 to Samsung's Invalidity Contentions)	
	13	U.S. Pub. No. 2007/0160243 to Dijkstra Claim Chart (Exhibit N12 to Samsung's Invalidity Contentions)	
	14	U.S. Pub. No. 2008/0130908 to Cohen Claim Chart (Exhibit N13 to Samsung's Invalidity Contentions)	
	15	U.S. Pub. No. 2001/0046304 to Rast Claim Chart (Exhibit N14 to Samsung's Invalidity Contentions)	
	16	U.S. Patent No. 8,270,629 to Bothra Claim Chart (Exhibit N15 to Samsung's Invalidity Contentions)	
	17	European Patent No. 1 519 625 to Victorian Claim Chart (Exhibit N16 to Samsung's Invalidity Contentions)	
	18	U.S. Patent No. 8,477,955 to Engle Claim Chart (Exhibit N17 to Samsung's Invalidity Contentions)	
	19	U.S. Patent No. 6,754,359 to Svean Claim Chart (Exhibit N18 to Samsung's Invalidity Contentions)	
	20	U.S. Patent No. 5,550,923 to Hotvet Claim Chart (Exhibit N19 to Samsung's Invalidity Contentions)	
	21	U.S. Patent No. 5,524,056 to Killion Claim Chart (Exhibit N20 to Samsung's Invalidity Contentions)	
	22	U.S. Pub. No. 2007/0092087 to Bothra Claim Chart (Exhibit N21 to Samsung's Invalidity Contentions)	

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	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	23	U.S. Patent No. 7,903,825 to Melanson Claim Chart (Exhibit N22 to Samsung's Invalidity Contentions)	
	24	U.S. Patent No. 5,251,263 to Andrea Claim Chart (Exhibit N23 to Samsung's Invalidity Contentions)	
	25	WO 93/26085 to Hohman Claim Chart (Exhibit N24 to Samsung's Invalidity Contentions)	
	26	U.S. Patent No. 8,189,803 to Bergeron Claim Chart (Exhibit N25 to Samsung's Invalidity Contentions)	
	27	U.S. Patent No. 7,936,885 to Frank Claim Chart (Exhibit N26 to Samsung's Invalidity Contentions)	
	28	U.S. Pub. No. 2007/0014423 to Darbut Claim Chart (Exhibit N27 to Samsung's Invalidity Contentions)	
	29	NACRE QuietPro Claim Chart (Exhibit N28 to Samsung's Invalidity Contentions)	
	30	Silynx QuietOps Claim Chart (Exhibit N29 to Samsung's Invalidity Contentions)	
	31	Aliph's Jawbone Claim Chart (Exhibit N30 to Samsung's Invalidity Contentions)	
	32	Bang & Olufsen's EarSet 2 Claim Chart (Exhibit N31 to Samsung's Invalidity Contentions)	
	33	Etymotic ER-6 Claim Chart (Exhibit N32 to Samsung's Invalidity Contentions)	

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	First Named Inventor	Goldstein et al.
	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	34	Gennum Zen Claim Chart (Exhibit N33 to Samsung's Invalidity Contentions)	
	35	Motorola H605 Claim Chart (Exhibit N34 to Samsung's Invalidity Contentions)	
	36	Peltor Lite-Com II Claim Chart (Exhibit N35 to Samsung's Invalidity Contentions)	
	37	Plantronics Discovery 655 Claim Chart (Exhibit N36 to Samsung's Invalidity Contentions)	
	38	Plantronic MX200 Series Claim Chart (Exhibit N37 to Samsung's Invalidity Contentions)	
	39	Sony S700 Claim Chart (Exhibit N38 to Samsung's Invalidity Contentions)	
	40	Motorola H5 Miniblue Claim Chart (Exhibit N39 to Samsung's Invalidity Contentions)	
	41	Sensaphonics 3D Active Ambient IEM Claim Chart (Exhibit N40 to Samsung's Invalidity Contentions)	
	42	U.S. Pub. No. 2005/0281422 to Armstrong Claim Chart (Exhibit N41 to Samsung's Invalidity Contentions)	
	43	U.S. Patent No. 6,424,721 to Hohn Claim Chart (Exhibit N42 to Samsung's Invalidity Contentions)	
	44	U.S. Pub. No. 2008/0063228 to Mejia Claim Chart (Exhibit N43 to Samsung's Invalidity Contentions)	

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	Art Unit	3992
	Examiner Name	Ge, Yuzhen
	Attorney Docket Number	3-00051-666EPR

	45	U.S. Patent No. 6,647,368 to Nemirovski Claim Chart (Exhibit N44 to Samsung's Invalidity Contentions)	
	46	U.S. Pub. No. 2007/0206825 to Thomasson Claim Chart (Exhibit N45 to Samsung's Invalidity Contentions)	
	47	U.S. Pub. No. 2009/0087003 to Zurek Claim Chart (Exhibit N46 to Samsung's Invalidity Contentions)	
	48	Isolated Word Recognition from In-Ear Microphone Data Using Hidden Markov Models (HMM) Claim Chart (Exhibit N47 to Samsung's Invalidity Contentions)	
	49	Audiometric Ear Canal Probe with Active Ambient Noise Control Claim Chart (Exhibit N48 to Samsung's Invalidity Contentions)	
	50	A Dual-Mode Human-Machine Interface for Robotic Control Based on Acoustic Sensitivity of the Aural Cavity Claim Chart (Exhibit N49 to Samsung's Invalidity Contentions)	

If you wish to add additional non-patent literature document citation information please click the Add button Add

EXAMINER SIGNATURE

Examiner Signature		Date Considered	

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through a citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

[1] See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. [2] Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). [3] For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. [4] Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. [5] Applicant is to place a check mark here if English language translation is attached.

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	Art Unit	3992
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CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).

OR

☐ That no item of information contained in the information disclosure statement was cited in a communication from a foreign patent office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of information contained in the information disclosure statement was known to any individual designated in 37 CFR 1.56(c) more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(2).

See attached certification statement.

☒ The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.

A certification statement is not submitted herewith.

SIGNATURE

A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.

Signature	/Jacob A. Snodgrass/	Date (YYYY-MM-DD)	2023-09-15
Name/Print	Jacob A. Snodgrass	Registration Number	66,032

This collection of information is required by 37 CFR 1.97 and 1.98. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 1 hour to complete, including gathering, preparing and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

Privacy Act Statement

The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these record s.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Electronic Acknowledgement Receipt

EFS ID:	48591373
Application Number:	90019169
International Application Number:	
Confirmation Number:	7859
Title of Invention:	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
First Named Inventor/Applicant Name:	11244666
Customer Number:	176134
Filer:	Jacob Andrew Snodgrass
Filer Authorized By:	
Attorney Docket Number:	0027071.00133
Receipt Date:	15-SEP-2023
Filing Date:	24-FEB-2023
Time Stamp:	08:54:25
Application Type:	Reexam (Third Party)

Payment information:

Submitted with Payment	no

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		90_019169_IDS_No_3_Cover_Letter_Corrected.pdf	694522 997362099cdef161deaec2e052753a6a4188128e	yes	2

	Multipart Description/PDF files in .zip description				
	Document Description	**Start**	**End**		
	Transmittal Letter	1	1		
	Reexam Certificate of Service	2	2		
Warnings:					
Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	IDS_Form_3.pdf	1318108 65093f6412ee3930ec0fc29e7e9b1073db508998	no	8
Warnings:					
Information:					
A U.S. Patent Number Citation or a U.S. Publication Number Citation is required in the Information Disclosure Statement (IDS) form for autoloading of data into USPTO systems. You may remove the form to add the required data in order to correct the Informational Message if you are citing U.S. References. If you chose not to include U.S. References, the image of the form will be processed and be made available within the Image File Wrapper (IFW) system. However, no data will be extracted from this form. Any additional data such as Foreign Patent Documents or Non Patent Literature will be manually reviewed and keyed into USPTO systems.					
3	Non Patent Literature	1_Samsung_Invalidity_Contentions_and_PR_3-3_and_3-4_Disclosures_excerpts.pdf	1450617 9b3226458f94f1d21e2061b9f2feaa703af39860	no	604
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4	Non Patent Literature	4-1_Exhibit-N1_Platz_077_Corrected_Reduced.pdf	10780006 2659caee0abc66d8722ad5bd865d3b29454627f3	no	104
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5	Non Patent Literature	4-2_Exhibit-N2_Kvaloy_Corrected.pdf	551310 3f8d19771a1a2ce23a7023d16f1e09c459d5bc27	no	79
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6	Non Patent Literature	4-3_Exhibit-N3_Inanaga_Corrected.pdf	594258 9e8829f8538959d379519755bb6ddeb2ff9ec93c	no	100
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Information:					

7	Non Patent Literature	4-4_Exhibit-N4_Rosenberg_Corrected.pdf	542765 b026d1652c92d59c10f71f62ddf7ceef083fc8c1	no	92
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8	Non Patent Literature	4-5_Exhibit-N5_Visser-958_Corrected.pdf	772289 279586007e347daed6ef21cc48ecc9ec0bb0fbf0	no	117
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9	Non Patent Literature	4-6_Exhibit-N6_Terlizzi_Corrected.pdf	766885 3c5c8c93298da2a1231f18dbdc81a4911574fc58	no	123
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10	Non Patent Literature	4-7_Exhibit-N7_Light_Corrected.pdf	856474 a3d17cf20f3448f36e9a2c8956cf6d818c5e44b3	no	103
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11	Non Patent Literature	4-8_Exhibit-N8_Boersma_Corrected.pdf	710413 bc1536d97441b213545f8ae174e53817060d6866	no	107
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12	Non Patent Literature	4-9_Exhibit-N9_McCune_Corrected.pdf	825729 880dbbf07459ade7c973fb6cf3126065fbcd7eb2	no	118
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13	Non Patent Literature	4-10_Exhibit-N10_Bose_Corrected.pdf	655410 0b6131ec7f0f5dfb03294783a3fbbf678a37a352	no	67
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14	Non Patent Literature	4-11_Exhibit-N11_Emoto_Corrected.pdf	811123 df7f63506671d6544e555649231470d6d8933fec	no	96
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15	Non Patent Literature	4-12_Exhibit-N12_Dijkstra-243_Corrected.pdf	685401 fcb1a74507ac41d54ef13b999ad033e939da3e21	no	117
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16	Non Patent Literature	4-13_Exhibit-N13_Cohen-908_Corrected.pdf	700664 92eb3f5b4107c48601db464b0ca9cc0735f65c82	no	115
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17	Non Patent Literature	4-14_Exhibit-N14_Rast_Corrected.pdf	714872 0e6fad65b91713a0f9f182496ee9e56766d79a0	no	123
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18	Non Patent Literature	4-15_Exhibit-N15_Bothra-629_Corrected.pdf	338807 73370c82e62439cf988bdab17e07eade1c7114f6	no	53
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19	Non Patent Literature	4-16_Exhibit-N16_Victorian-625_Corrected.pdf	965549 9947e6cf8e9fecd8b84d1c2be6e00844b2c3f7c1	no	139
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20	Non Patent Literature	4-17_Exhibit-N17_Engle_Corrected.pdf	593235 d5f5337d930e4eca7f8566555f89a956ddba633b	no	90
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21	Non Patent Literature	4-18_Exhibit-N18_Svean-359_Corrected.pdf	678046 bfa17655a858b2b2ed9e0bc0af972cd61b3b3404	no	128
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22	Non Patent Literature	4-19_Exhibit-N19_Hotvet_Corrected.pdf	474565 fdf69d94e4dc5aa5718b11c7450598a1aba93455	no	73
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23	Non Patent Literature	4-20_Exhibit-N20_Killion-056_Corrected.pdf	516470 4d616da4f8a56047a0d3abd397118a9e039f04e6	no	74
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24	Non Patent Literature	4-21_Exhibit-N21_Bothra-087_Corrected.pdf	481305 f7024de90d17b8653f1c850f787907df83016181	no	72
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25	Non Patent Literature	4-22_Exhibit-N22_Melanson_Corrected.pdf	626779 e9dbda64464fc6ef3764070a56708107d5d272c2	no	86
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26	Non Patent Literature	4-23_Exhibit-N23_Andrea_Corrected.pdf	665721 8e90319b25b95e7775b07bc35336f8990700bcf4	no	90
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27	Non Patent Literature	4-24_Exhibit-N24_Hohman_Corrected.pdf	534691 25593e12fe9a14ff21a2ee7f691c70cbc4c6cc49	no	82
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28	Non Patent Literature	4-25_Exhibit-N25_Bergeron_Corrected.pdf	812689 e0fcd9a8ca7c7f09d8b6763153bb859e9e427868	no	100
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29	Non Patent Literature	4-26_Exhibit-N26_Frank_Corrected.pdf	632152 08f723b7728f5308e19b81ef781d17f9ef01aa22	no	116
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30	Non Patent Literature	4-27_Exhibit-N27_Darbut-423_Corrected.pdf	898156 890d5502c091a039b0bb736827e555e989e96ef2	no	92
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31	Non Patent Literature	4-28_Exhibit-N28_QuietPro_Corrected.pdf	2348076 52e9b3308f451e3b72d0f7b22d24eb84325017e2	no	205
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32	Non Patent Literature	4-29_Exhibit-N29_QuietOps_Corrected.pdf	2086947 0bceff01f7bb6a5cafe8486f50038179c962bdec	no	172
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33	Non Patent Literature	4-30_Exhibit-N30_Jawbone_Corrected.pdf	5026649 e945719074aca475f40c3f86faa299de4dbe5e8b	no	189
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34	Non Patent Literature	4-31_Exhibit-N31_EarSet_2_Corrected.pdf	3115555 aee2f971e6181cab90c6142072de9018bce5240a	no	172
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35	Non Patent Literature	4-32_Exhibit-N32_Etymotic_ER-6_Corrected.pdf	3903989 783c10aaf0b792db6cda51849ae313890e4c2111	no	256
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36	Non Patent Literature	4-33_Exhibit-N33_Zen_Corrected.pdf	7137622 c08c530f290edc243198ce9b29407e7df61d4200	no	188
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37	Non Patent Literature	4-34_Exhibit-N34_Motorola_H605_Corrected.pdf	2219814 3e97b11ec239f6c870a5ab4749a9fc7070796a92	no	121
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38	Non Patent Literature	4-35_Exhibit-N35_Peltor_Lite-Com_II_Corrected.pdf	8042959 835f1c128cf0064c02556eaa58982fa9707afafb	no	223
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39	Non Patent Literature	4-36_Exhibit-N36_Discovery_655_Corrected.pdf	6514152 7b56870efbb668357d615222840fa5ef50242f87	no	308
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40	Non Patent Literature	4-37_Exhibit-N37_MX200_Series_Corrected.pdf	2183251 b3aaeffad482d741a558c3d1210135c7713dbeeb	no	121
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41	Non Patent Literature	4-38_Exhibit-N38_Sony_S700_Corrected.pdf	2799391 726dacb03bc986047d195f6a88bad37dec1086e6	no	155
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42	Non Patent Literature	4-39_Exhibit-N39_H5_Miniblue_Corrected.pdf	2764000 727e18619596b7845cd36360a51b76a6f77a230c	no	172
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43	Non Patent Literature	4-40_Exhibit-N40_3D_Active_Ambient_IEM_Corrected.pdf	9112979 305b0ed9acc72ab2fbb97f4e407c7f871b4bd69b	no	291
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44	Non Patent Literature	4-41_Exhibit-N41_Armstrong-422_Corrected.pdf	1212908 b8ad71d2288713be5ab1d0c66507664cf0bdc767	no	232
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45	Non Patent Literature	4-42_Exhibit-N42_Hohn_Corrected.pdf	729121 0ac9a4a23db9ff2e3c9141e846fab8cd27d52360	no	160
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46	Non Patent Literature	4-43_Exhibit-N43_Mejia-228_Corrected.pdf	1594147 a17ce150678146341 06ca1ffe4ceeeda4ec3fca8	no	180
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47	Non Patent Literature	4-44_Exhibit-N44_Nemirovski-368_Corrected.pdf	1730412 bcca11956a4ebbe87754e4075201b6d91eb3ef80	no	264
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48	Non Patent Literature	4-45_Exhibit-N45_Thomasson_Corrected.pdf	519172 5f20dfce505ac35d6b7d92986f40a5f8b740eb34	no	95
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49	Non Patent Literature	4-46_Exhibit-N46_Zurek-003_Corrected.pdf	1062363 15233f2f12a7a3e099193c74621d1c6ffb99a629	no	230
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50	Non Patent Literature	4-47_Exhibit-N47_Kurcan_Corrected.pdf	12416364 3cfb4a3f81f5fc92324ac9db867c36a274745854	no	444
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51	Non Patent Literature	4-48_Exhibit-N48_Rafaely_Corrected.pdf	2356621 a1299eb7ee1b6c84b2aab0140e66f139044f951f	no	205
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52	Non Patent Literature	4-49_Exhibit-N49_Vaidyanathan_Corrected.pdf	6353765 db35f3f22fb60760b86007b75f5e9fdfa240a63f	no	155
Warnings:					
Information:					
		Total Files Size (in bytes):	116879268		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Control No.	90/019,169
U.S. Patent No.	11,244,666
Reexamination Filing Date	February 24, 2023
Inventors	Goldstein et al.
Title	METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION
Examiner	Ge, Yuzhen
Art Unit	3992
Confirmation No.	7859

Submitted via EFS-Web
Mail Stop Ex Parte Reexamination
Attn: Central Reexamination Unit
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

TRANSMITTAL LETTER FOR INFORMATION DISCLOSURE STATEMENT

Patent Owner herewith submits an Information Disclosure Statement (form PTO/SB/08a).

Patent Owner submits herewith the IDS fee. Should any additional fees be due in connection with this paper or at any other time during this proceeding, the Commissioner is authorized to charge such fees to Deposit Account No. 60-3815.

Dated: September 15, 2023

Respectfully submitted,

/s/ Jacob A. Snodgrass
Jacob A. Snodgrass
Registration No. 66,032

PV LAW LLP
Customer No. 176134
5335 Wisconsin Avenue, N.W., Suite 440
Washington, D.C. 20015
T: (202) 371-6861
F: (202) 888-3163

CERTIFICATE OF SERVICE

Pursuant to 37 C.F.R. § 1.550(f), the undersigned hereby certifies that on September 15, 2023, a copy of this letter as well as the Information Disclosure Statement (form PTO/SB/08a) and cited non-patent literature documents (on flash drive) were served on Third Party Requester at the following address.

Ali R. Sharifahmadian
Arnold & Porter Kaye Scholer LLP
601 Massachusetts Ave., N.W.
Washington, DC 20001

Dated: September 15, 2023

Respectfully submitted,

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US011244666C1

(12) EX PARTE REEXAMINATION CERTIFICATE (12425th)
United States Patent
Goldstein et al.

(10) Number: US 11,244,666 C1
(45) Certificate Issued: Oct. 12, 2023

(54) METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

(71) Applicant: **Staton Techiya LLC**, Delray Beach, FL (US)

(72) Inventors: **Steven Wayne Goldstein**, Delray Beach, FL (US); **John Usher**, Beer (GB); **Marc Andre Boillot**, Plantation, FL (US)

(73) Assignee: **STATON TECHIYA, LLC**, Delray Beach, FL (US)

Reexamination Request:
No. 90/019,169, Feb. 24, 2023

Reexamination Certificate for:

Patent No.:	**11,244,666**
Issued:	**Feb. 8, 2022**
Appl. No.:	**16/987,396**
Filed:	**Aug. 7, 2020**

Related U.S. Application Data

(63) Continuation of application No. 16/669,490, filed on Oct. 30, 2019, now Pat. No. 10,810,989, which is a continuation of application No. 16/193,568, filed on Nov. 16, 2018, now Pat. No. 10,535,334, which is a continuation of application No. 14/574,589, filed on Dec. 18, 2014, now Pat. No. 10,134,377, which is a continuation of application No. 12/017,878, filed on Jan. 22, 2008, now Pat. No. 8,917,894.

(60) Provisional application No. 60/885,917, filed on Jan. 22, 2007.

(51) **Int. Cl.**

G10K 11/178	(2006.01)
H04R 3/00	(2006.01)
H04R 1/10	(2006.01)
G10K 11/00	(2006.01)
H04R 29/00	(2006.01)

(52) **U.S. Cl.**
CPC ***G10K 11/17827*** (2018.01); ***G10K 11/002*** (2013.01); ***H04R 1/1016*** (2013.01); ***H04R 1/1083*** (2013.01); ***H04R 3/002*** (2013.01); ***H04R 3/005*** (2013.01); ***H04R 29/001*** (2013.01); *H04R 2225/41* (2013.01); *H04R 2460/05* (2013.01); *H04R 2499/11* (2013.01)

(58) **Field of Classification Search**
CPC .. H04R 29/001; H04R 1/1083; H04R 1/1016; H04R 3/002; H04R 3/005; H04R 2499/11; H04R 2225/41; H04R 2460/05; G10K 11/17827; G10K 11/002
See application file for complete search history.

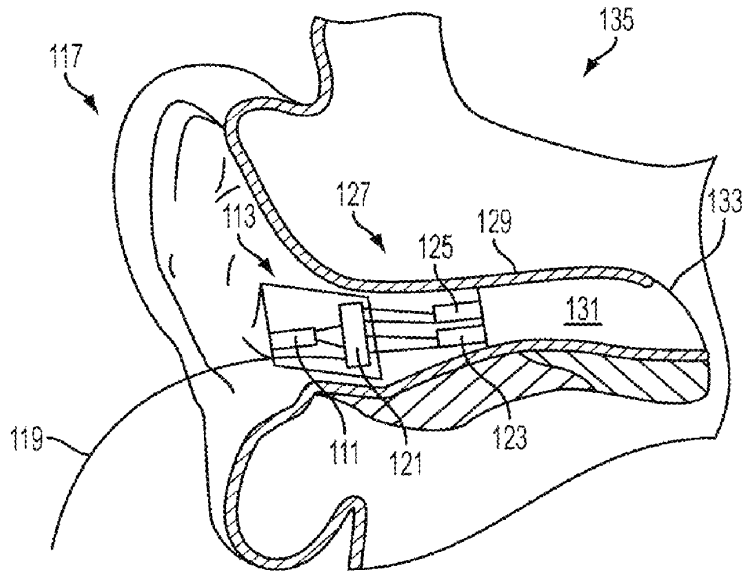
(56) **References Cited**

To view the complete listing of prior art documents cited during the proceeding for Reexamination Control Number 90/019,169, please refer to the USPTO's Patent Electronic System.

Primary Examiner — Yuzhen Ge

(57) **ABSTRACT**

Earpieces and methods for acute sound detection and reproduction are provided. A method can include measuring an external ambient sound level (xASL), monitoring a change in the xASL for detecting an acute sound, estimating a proximity of the acute sound, and upon detecting the acute sound and its proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece. Other embodiments are disclosed.



EX PARTE REEXAMINATION CERTIFICATE

NO AMENDMENTS HAVE BEEN MADE TO THE PATENT

AS A RESULT OF REEXAMINATION, IT HAS BEEN DETERMINED THAT:

The patentability of claims **1-11** is confirmed.

* * * * *