4kHzの周波数部分で約400Hzの間隔の規則的な凹凸特性(ピークディップ)が生じており、距離Dが1mの場合ではやや不規則な凹凸特性となり、距離Dが2mの場合では周波数間隔が異なる複数の凹凸特性が組み合わされた特性となっている。また、距離Dが同一の場合は左右でほぼ同一の特性が得られている。

[0136] パーソナルコンピュータ61は、生成したインパルス音の測定信号と増幅器66a, 66bから入力されるインパルス応答信号の波形とを比較して、距離Dごとの頭部伝達特性を求める。距離Dごとに求めた頭部伝達特性が音源を仮想的に距離Dの位置に定位させて、その仮想的な音源からの音声信号を受聴者に受聴させるための特性である。なお、本実施形態では距離Dを0.5m, 1m, 2mとしたが、さらに多くの距離においてそれぞれの特性を求めてもよく、距離Dの間隔を0.5mよりも狭くしてもよい。

[0137] 以上のようにして得られた頭部伝達特性は図31の伝達関数メモリ334に記憶される。ズーム倍率検出器331で検出されたズーム倍率に対して、記憶された伝達関数のいずれを用いるかは、一例として、カメラ部11における図示しない自動焦点測定機能により計測される被写体までの距離をズーム倍率で除して得られる係数αを用いる。例えば被写体までの距離が10mでズーム倍率が5の場合、係数αは2となる。被写体までの距離が10mでズーム倍率が10の場合、係数αは1となり、ズーム倍率が20の場合には係数αは0.5となる。

[0138] 図40を用いて、音声ズーム処理部33aでの動作を含む映像音声記録再生装置102の動作についてさらに説明する。図40において、ステップS211にて、加算器335はバイノーラルマイク3からの左右チャンネルのバイノーラル音声信号を加算して加算信号Sを得る。ステップS212にて、ズーム倍率検出器331は操作部48の操作に応じて制御部47で得られるズーム倍率を検出する。ステップS213にて、係数算出器332はズーム倍率に応じて伝達関数メモリ334に記憶された複数の伝達関数の内のいずれを選択するかを決めたり、可変増幅器337での増強度を表す係数αを算出したりする。なお、係数αとしては被写体までの距離をズーム倍率で除して得られる数をそのまま用いてもよいし、被写体までの距離をズーム倍率で除して得られる数に基づいて生成した値であってもよい。

[0139] ステップS214にて、関数取得器333は係数αに応じて伝達関数メモリ334より伝

達関数を取得し、畳み込み演算器336は加算器335から出力された加算信号に伝達関数を畳み込む。ステップS215にて、可変増幅器337は畳み込み演算器336の出力信号に係数αを乗じて増強する。ステップS216にて、加算器338a，338bは、左，右チャンネルのバイノーラル音声信号と可変増幅器337の出力信号とを加算する。そして、ステップS216にて、ズームアップ音声信号を記録媒体44に記録する。ステップS218にて、制御部47は記録が終了したか判定し、終了していなければ(NO)、ステップS211から繰り返す。ステップS218で記録が終了していれば(YES)、音声ズーム処理部33aでの処理を終了する。

[0140] ≪第3の実施形態≫

第2の実施形態は音声のズームアップ処理を記録側で行うものであるが、第3の実施形態は音声のズームアップ処理を再生側で行うものである。図41に示す第3の実施形態の映像音声記録再生装置103において、図3に示した第1の実施形態の映像音声記録再生装置101と同じ機能を有する構成部分には同一符号を付し、その説明を省略する。映像音声記録再生装置103は映像音声記録再生装置101と比較して、音声ズーム処理部33bを分離化部15の後段に設けており、ズーム倍率検出器331を多重化部13の前段に設けている点が相違している。なお、図41では、ヘッドフォン55とヘッドフォン55の接続端子である音声出力端子37cの図示を省略している。

[0141] 映像音声記録再生装置103の動作について説明する。操作部48が操作され、制御部47で生成されるカメラ部11のレンズ駆動用信号はカメラ部11とズーム倍率検出器331に供給される。ズーム倍率検出器331は、レンズ駆動用信号のズーム方向，ズーム速度，レンズ駆動時間を解析してズーム倍率を検出する。検出されたズーム倍率を示すズーム倍率情報は多重化部13に供給される。多重化部13は符号化映像信号と符号化音声信号とバイノーラルフラグ信号に併せてズーム倍率情報も多重化する。記録・再生部14はズーム倍率情報も含む多重化信号を記録媒体44に記録する。

[0142] 記録・再生部14は記録媒体44に記録された多重化信号を再生し、分離化部15は符号化映像信号と符号化音声信号とバイノーラルフラグ信号に併せてズーム倍率情

報も分離する。ズーム倍率情報は音声ズーム処理部33bに入力される。

[0143] 図42は音声ズーム処理部33bの具体的な構成例である。図42に示すように、音声ズーム処理部33bは図29の音声ズーム処理部33と比較して、ズーム倍率検出器331が省かれている点と加算器338a，338bに入力される信号がクロストークキャンセラ27の出力信号である点で相違している。

[0144] 図42において、係数算出器332は分離化部15で分離されて入力されたズーム倍率情報を用いて可変増幅器337で入力信号を増幅する際に用いる係数αを算出する。加算器335はオーディオ復号化部26から入力されるバイノーラル音声信号を加算する。可変増幅器337は加算器335からの出力信号を係数算出器332から入力された係数αに応じて増幅する。加算器338a，338bはクロストークキャンセラ27からの出力信号と可変増幅器337からの増幅信号とを加算する。

[0145] ところで、カメラ部11におけるズーム処理を例えばDSPを用いてソフトウェアによって演算処理して実行させる場合がある。音声ズーム処理を再生時に行う場合にはズーム処理に必要なDSPの信号処理時間を確保する必要がない。そのため、記録時にDSPによって撮影した映像信号の最適化処理や映像信号の符号化、または記録制御などの信号処理を十分に行うことができる。音声ズーム処理を再生時に行うことにより、DSPでの演算工数がズーム処理のための演算処理に割り当てられて、記録を行うための演算時間が不足するという不具合が発生するおそれを回避することができる。

[0146] 図43は、図42の音声ズーム処理部33bに対して、図31と同様、正中面の音声信号に近接効果を与える頭部伝達関数を畳み込むように構成した音声ズーム処理部33cを示している。音声ズーム処理部33cは音声ズーム処理部33bと比較して、関数取得器333、伝達関数メモリ334、畳み込み演算器336が追加されている点で相違している。関数取得器333、伝達関数メモリ334、畳み込み演算器336の動作は図31と同様であり、説明を省略する。

[0147] ≪第4の実施形態≫

図44に示す第4の実施形態の映像音声記録再生装置104は、音声のズームアップ処理を記録媒体44の再生時に外部操作により手動で行うよう構成したものである。

即ち、ズーム倍率情報が記録媒体44に記録されていない場合に、視聴者59がモニタ52で再生した映像信号を見ながら音声のズームアップ処理を行うものである。視聴者59による手動で行う音声のズームアップ処理をマニュアル音声ズーム処理と称することとする。

[0148] 図44に示す映像音声記録再生装置104は、映像音声記録再生装置103と比較して、ズーム倍率検出器331が省略され、音声ズーム処理部33bの代わりに音声ズーム処理部33dを備えている点で相違している。

[0149] 視聴者59が操作部48を操作してマニュアル音声ズームを行うよう指示すると、制御部47はズームアップ制御信号を音声ズーム処理部33dに入力される。音声ズーム処理部33dはズームアップ制御信号に従ってオーディオ復号化部26で復号されたバイノーラル音声信号に対してズームアップ処理を施す。

[0150] 図45は音声ズーム処理部33dの具体的な構成例である。図45に示すように、音声ズーム処理部33dは、図42のズーム処理部33bと比較して、制御部47からのズームアップ制御信号が入力されるズーム倍率検出器331aが設けられ、係数算出器332には分離化部15からのズーム倍率情報ではなくズーム倍率検出器331aで生成したズーム倍率情報を入力している点で相違している。その他の部分はズーム処理部33bと同様の動作であり、説明を省略する。

[0151] 図46を用いて第4の実施形態におけるマニュアル音声ズーム処理についてさらに説明する。図46において、ステップS221にて、加算器335は再生して得られたバイノーラル音声信号の左右の信号を加算して加算信号Sを得る。ステップS222にて、制御部47は操作部48によって音声ズーム倍率が可変されたか否かを判定する。ステップS222で音声ズーム倍率が可変された場合は(YES)、ステップS223に移り、可変されなかった場合は(NO)、ステップS226に移る。

[0152] 音声ズーム倍率が可変された場合にはステップS223にて、ズーム倍率検出器331aはズームアップ制御信号に基づいてズーム倍率を算出する。ステップS224にて、係数算出器332はズーム倍率検出器331aから入力されたズーム倍率に基づいて係数αを算出する。係数αは、音源を視聴者の前方に定位させるための頭部伝達関数の特性を含んでいてもよい。ステップS225にて、係数αの値を新たに算出された

値に更新する。

[0153] そして、ステップS226にて、可変増幅器337は加算信号Sに係数αを乗算してαSを生成する。なお、ステップS223～S225を経ていない場合は、係数αは音声ズーム倍率が可変する前の値である。ステップS227にて、加算器338a，338bはクロストークキャンセラ27によってクロストークキャンセル処理が施されたバイノーラル音声信号に信号αSを加算する。ステップS228にて、スイッチSw2及び音声出力端子37bを介してステップS227で得られた音声信号を出力する。ステップS229にて、制御部47は再生が終了したか否かを判定し、終了していなければ(NO)、ステップS221に戻り、終了していれば(YES)、処理を終了する。

[0154] ≪第5の実施形態≫

図47に示す第5の実施形態の映像音声記録再生装置105は、ズームアップ音声信号をヘッドフォン55で受聴するのに好適な構成を示している。図47に示す映像音声記録再生装置105は、図41の映像音声記録再生装置103と比較して、音声ズーム処理部33bの代わりに音声ズーム処理部33eを備えており、音声ズーム処理部33eから出力された音声信号を音声出力端子37cを介してヘッドフォン55へと供給している点で相違している。ヘッドフォン55には、クロストークキャンセラ27によってクロストークキャンセル処理を施しておらず、音声ズーム処理部33eによってズームアップ処理を施したバイノーラル音声信号を供給する。

[0155] 図48は音声ズーム処理部33eの具体的な構成例である。音声ズーム処理部33eは図42の音声ズーム処理部33bと比較して、加算器338c，338dが追加されている点で相違している。加算器338c，338dは、オーディオ復号化部26で復号して得られたバイノーラル音性信号に可変増幅器337から出力されたズームアップ音声信号を加算する。加算器338c，338dによって加算されたヘッドフォン受聴用音声信号は音声出力端子37cを介してヘッドフォン55へと供給される。

[0156] 以上説明した第2～第5の実施形態によるズームアップ音声信号の再生効果は以下の通りである。

[0157] カメラ部11の視野角がワイドに設定され、ズーム倍率が小さい場合には加算器335で加算された信号は可変増幅器337で増強されないため、視聴者59は、モニタ52

に表示された映像信号と撮影者300を囲む360度の音声信号をスピーカ53，54で再生した臨場感を有する音声信号とを視聴することになる。視野角をワイドに設定した場合でも視野角は60度程度である。視聴者59は、映像の視野角と音声信号を集音した際の角度範囲との違いにより、モニタ52に表示される被写体から発する音が不足するいわゆる中抜け音声であると認識する場合がある。これに対して、前述の如く音声ズームアップ処理が施された音声信号は、撮影者300に対して正中面からの信号成分を増強してバイノーラル音声信号に加算するため、中抜けが補償された音声信号となる。よって、視聴者59が中抜け音声であると認識することを防止し、視聴者59は第1実施例と比較してさらに違和感の少ない臨場感のある音声を受聴することが可能となる。

[0158] ≪第6の実施形態≫

図49，図50に示す第6の実施形態の映像音声記録再生装置106は、第1～第5の実施形態のように内蔵ステレオマイク21とバイノーラルマイク3とを別々に設けたものではなく、通常のステレオ集音用のマイクとバイノーラルマイクとを共用させた構成を示している。図49は第6の実施形態も映像音声記録再生装置106の外観構成例を示す平面図であり、図50は映像音声記録再生装置106の具体的な内部構成例を示すブロック図である。図49，図50において、図1，図3と同一の機能を有する構成部分には同一符号を付し、その説明を省略する。

[0159] 図49に示すように、映像音声記録再生装置106は上面にマイク31e，31fを載置するためのマイク載置部35a，35bとマイク31e，31fに接続されたマイクコード310e，310fを収納するコード収納部34を備えている。

[0160] 図49において、マイク31e，31fを用いて通常のステレオ音声にて集音する場合は、撮影者300はマイク31e，31fをマイク載置部35a，35bに載置して集音する。バイノーラル音声の集音を行う場合は、撮影者300はコード収納部34からマイクコード310e，310fを引き出してマイク31e，31fを耳302に装着して集音する。映像音声記録再生装置106はマイク載置部35a，35bに載置されるマイク31e，31fを検出する図示しない突起状検出部を有する。突起状検出部に連動して動作するスイッチ（図50のスイッチSw4に相当）のオン／オフにより、映像音声記録再生装置106はマイク3

1e, 31fがマイク載置部35a, 35bに載置されているか否かを検出する。マイク載置部35a, 35bに載置されているか否かの検出はこれに限定されることはなく、例えばマイク31e, 31fに内蔵されている永久磁石から発生する磁界をホール素子や磁気抵抗素子を用いて検出するようにしてもよい。

[0161] 具体的には、図50において、スイッチSw4は、マイク31e, 31fがマイク載置部35a, 35bに載置されていなければ端子eに接続してオフ状態となり、マイク31e, 31fが載置されていれば端子fに接続してオン状態となる。載置検出部41aは、スイッチSw4が端子eに接続しているか端子fに接続しているかによってマイク31e, 31fがマイク載置部35a, 35bに載置されているか否かを検出する。載置検出部41aによる検出信号は制御部47へ供給される。

[0162] 載置検出部41aが、マイク31e, 31fが載置されていることを検出した場合には、マイク31e, 31fは通常のステレオ集音を行い、制御部47は、映像音声記録再生装置106がノーマルモードでの撮影に対応した記録動作を行うよう回路各部を制御する。この場合、マイク31e, 31fの役割は図3における内蔵ステレオマイク21と等価であり、フラグ情報生成部42はバイノーラルモードであることを示すバイノーラルフラグ信号を生成しない。一方、載置検出部41aが、マイク31e, 31fが載置されていないことを検出した場合には、撮影者300がマイク31e, 31fを耳302に装着してバイノーラル集音するバイノーラルモードであるとみなし、制御部47は、映像音声記録再生装置106がバイノーラルモードでの撮影に対応した記録動作を行うよう回路各部を制御する。この場合、フラグ情報生成部42は制御部47による制御に基づいてバイノーラルフラグ信号を生成する。

[0163] 第6の実施形態では、マイク31e, 31f、マイク載置部35a, 35b、載置検出部41a、制御部47が全体として、周囲音を集音するマイクとして、撮影者の耳に装着するバイノーラルマイクを使用するかバイノーラルマイク以外のマイクを使用するかを切り替える切り替え部として動作している。

[0164] ここで、図51を用いてコード収納部34の構造の一例について説明する。図51(A)はコード収納部34を上面側から見た内部構造であり、マイクコード310e, 310fは回転軸343を有するリール341に巻き付けられている。図51(B)はコード収納部34を

下面側から見た内部構造であり、リール341の内部には渦巻状のスプリング342が備えられている。マイク31e，31fがマイク載置部35a，35bに載置されているか否かを検出する代わりに、またはこれに加えて、リール341の回転角を検出することによりバイノーラルモードであるか否かを検出してもよい。

[0165] ≪第7の実施形態≫

図52に示す第7の実施形態の映像音声記録再生装置107は、集音して得られた音声信号を装置本体に無線伝送する無線型バイノーラルマイクを用いた構成を示している。図52において、図1と同一の機能を有する構成部分には同一符号を付し、その説明を省略する。

[0166] 図52において、映像音声記録再生装置107は、図1の外部マイク接続端子32の代わりに無線送受信部39を備え、バイノーラルマイク3の代わりに無線型バイノーラルマイク38を用いて音声収録を行う。撮影者300は、装置本体と無線結合される無線型バイノーラルマイク38を頭部に装着し左右のマイク38a，38bを耳302に挿入して集音することにより、マイクコードに邪魔されることなく撮影を行うことができる。また、撮影者300と、図示しない集音者の2人による撮影も可能となる。

[0167] 図53を用いて映像音声記録再生装置107の内部構成について説明する。図53において、図3と同一の機能を有する構成部分には同一符号を付し、その説明を省略する。図53に示す映像音声記録再生装置107は、図3の映像音声記録再生装置101と比較して、外部マイク接続端子32及び接続検出部41の代わりに無線送受信部39を備えている点で相違している。

[0168] 制御部47は、無線型バイノーラルマイク38が装置本体から所定の距離にあると判断し、無線送受信部39が無線型バイノーラルマイク38からバイノーラル音声信号を受信する場合に、スイッチSw1を端子bに接続させて無線型バイノーラルマイク38からバイノーラル音声信号をオーディオ符号化部22に供給させる。この場合、制御部47は、フラグ情報生成部42がバイノーラルフラグ信号を生成するよう制御する。また、制御部47は、無線型バイノーラルマイク38が装置本体から所定の距離を越えると判断した場合には、スイッチSw1を端子aに接続させ、内蔵ステレオマイク21からのステレオ音声信号をオーディオ符号化部22に供給させる。この場合、フラグ情報生成

部42はバイノーラルフラグ信号を生成しない。

[0169] 図54に無線型バイノーラルマイク38及び無線送受信部39の内部構成例を示し、それらの動作について説明する。

[0170] 図54に示すように、無線型バイノーラルマイク38のマイク38aは、マイクユニット381、マイクアンプ382、送受信ユニット383、アンテナ384、警告信号発信部385を備えて構成されている。図示を省略しているが、マイク38bは、警告信号発信部385を備えていない点以外はマイク38aと同様の構成である。無線送受信部39は、送受信ユニット391、マイク判定部392、距離測定部393、通信可能圏内判定部394、警告信号発信部395、アンテナ396を備えて構成されている。

[0171] マイク38a(38b)のマイクユニット381はバイノーラル音声信号を生成する。マイクアンプ382はマイクユニット381からのバイノーラル音声信号を増幅する。送受信ユニット383はマイクアンプ382からの増幅されたバイノーラル音声信号を所定の変調方式により変調してアンテナ384を介して送信する。警告信号発信部385は後述する無線送受信部39の警告信号発信部395で生成され、送受信ユニット391及び送受信ユニット383を介して伝送された警告信号を基に警告信号を発生する。

[0172] 無線送受信部39のアンテナ396は左右のマイク38a, 38bから送信される変調信号を受信する。送受信ユニット391は受信した変調信号を復調してバイノーラル音声信号を生成すると共に、変調信号の受信電力の強度を測定する。距離測定部393は測定された受信電力の強度を基に無線送受信部39から無線型バイノーラルマイク38までの距離を推定する。通信可能圏内判定部394は推定された距離が通信可能な所定の距離以内であるか否かを判定する。通信可能圏内判定部394の判定結果は制御部47に供給される。制御部47は、推定された距離が通信可能な所定の距離以内であればスイッチSw1を端子bに接続させ、フラグ情報生成部42がバイノーラルフラグ信号を生成するよう制御する。制御部47は、推定された距離が通信可能な所定の距離を超える場合にはスイッチSw1を端子aに接続させる。

[0173] 警告信号発信部395は、通信可能圏内判定部394で推定された距離が通信可能な所定の距離を超えると判定した場合に警告信号を生成する。警告信号は制御部47に供給される。制御部47は警告マークを生成して表示部17に供給して、表示部17

に所定の警告マークを表示させる。マイク判定部392は警告信号発信部395で警告信号を生成していない場合は正常にバイノーラル音声信号が得られていると判定し、送受信ユニット391にて復調して得られたバイノーラル音声信号を、スイッチSw1を介してオーディオ符号化部22に供給する。

[0174] 図55に、無線型バイノーラルマイク38及び映像音声記録再生装置107でなされる警告表示の例を示す。マイク38aには発光ダイオード(LED)386を先端部に装着した棒状部材が設けられている。LED386には警告信号発信部385で生成された警告信号が供給され、警告信号に基づいてLED386が点滅(または点灯)する。LED386が点滅に加えて、またはLED386が点滅の代わりに警告音を発生させてもよい。この場合には、警告音がマイクユニット381で受聴されないよう(受聴されにくいよう)、警告音のレベルを小さくしたり、警告音の周波数を例えば数十Hz以下の低周波数としたりすることが好ましい。

[0175] 警告信号発信部395は、無線型バイノーラルマイク38が破線の円で示す通信可能圏を越えたと判定した場合に警告信号を生成する。図55に示すように、無線型バイノーラルマイク38が通信可能圏を越えていれば、表示部17に所定の警告マークが表示される。

[0176] 図56は表示部17に表示される警告マークの例を示している。図56(A)に示す警告マーク171aは、図5(A)に示したバイノーラルマーク171上に×印を重ねて表示したものである。図56(B)に示す警告マーク172aは、図5(B)に示すマーク172をグレー表示したものである。警告マークとしては図56(A),(B)いずれのマークを用いてもよく、勿論これ以外のマークを用いてもよい。警告表示を行ったにもかかわらず無線送受信部39での受信電力が受信限界以下となると想定される場合には、制御部47は無線型バイノーラルマイク38から内蔵ステレオマイク21へと切り替える。

[0177] 図57を用いて映像音声記録再生装置107の動作についてさらに説明する。図57において、ステップS251にて、制御部47はバイノーラルモードであるか否かを判定する。ステップS251でバイノーラルモードであると判定されなければ(NO)、ステップS253にて、記録・再生部14は内蔵ステレオマイク21で集音して通常のステレオ音声信号を記録媒体44に記録する。ステップS251でバイノーラルモードであると判定

されれば(YES)、ステップS252にて無線送受信部39は無線型バイノーラルマイク38からの送信信号を受信する。ステップS254にて、距離測定部393は受信電力の強度(受信強度)に基づいて無線送受信部39から無線型バイノーラルマイク38までの距離を検出する。ステップS255にて、通信可能圏内判定部394は検出した距離が所定の距離以内であるか否かを判定する。

[0178] ステップS255で所定の距離以内であると判定されなければ(NO)、ステップS257にて、制御部47は、警告表示時間tが0(非提示)であるか否か判定し、警告表示時間tが0であればステップS300にて警告信号を発生するよう警告信号発信部395を制御する。警告信号の発生後、ステップS254に戻り同じ処理を繰り返す。ステップS257にて、警告表示時間tが0でなければ(NO)、ステップS258にて、制御部47は、警告表示時間tが予め定めた最大時間tmaxを越えたか否か判定し、最大時間tmaxを越えていなければ(NO)、ステップS300に移り、同様にテップS254に戻る。最大時間tmaxを越えていれば(YES)、ステップS259にて、制御部47は、無線型バイノーラルマイク38から内蔵ステレオマイク21へと切り替えるようスイッチSw1を制御し、警告信号の発生を停止するよう警告信号発信部395を制御する。そして、ステップS253に移る。

[0179] ステップS255で所定の距離以内であると判定されれば(YES)、ステップS256にて、制御部47は、警告信号発信部395が警告信号を発生中であれば、警告信号の発生を停止するよう警告信号発信部395を制御する。そして、ステップS301にて、記録・再生部14は無線型バイノーラルマイク38で集音してバイノーラル音声信号を記録媒体44に記録する。ステップS302にて、制御部47は、記録終了の操作がなされたか否か判定し、記録終了の操作がなされていなければ(NO)、ステップS251に戻り、記録終了の操作がなされていれば(YES)、処理を終了する。

産業上の利用可能性

[0180] 本発明は、民生用のビデオカメラのみならず、業務用のビデオカメラにおいても、撮像した映像と共に臨場感のある音声を再生することが必要な映像音声記録再生装置に適用できる。また、動画撮影機能を有するデジタルカメラや携帯電話にも適用できる。さらには、本発明は映像信号と音声信号との双方を記録再生する映像音声記録

再生装置に用いて好適であるが、音声信号のみを記録再生する音声記録再生装置に用いても十分に効果的である。

請求の範囲

[1] 被写体を撮像した映像信号と、前記被写体から発せられた音を含む撮影者(300)の周囲音を集音した音声信号とを記録する映像音声記録装置(101, 102, 103, 104, 105, 107)において、

前記被写体を撮像するカメラ部(11)と、

前記周囲音を集音するマイクとして、前記撮影者(300)の耳に装着するバイノーラルマイク(3)を使用するかバイノーラルマイク(3)以外のマイクを使用するかを切り替える切り替え部(Sw1)と、

前記カメラ部(11)より出力された映像信号を処理するビデオ処理部(12)と、

前記周囲音を集音するマイクより出力された音声信号を処理するオーディオ処理部(22)と、

前記切り替え部(Sw1)によって前記周囲音を集音するマイクとして前記バイノーラルマイク(3)に切り替えた場合に、前記周囲音を集音する際の音声モードをバイノーラルモードとして、バイノーラルモードであることを示すバイノーラルフラグ信号を生成するフラグ情報生成部(42)と、

前記ビデオ処理部(12)によって処理された映像信号と前記オーディオ処理部(22)によって処理された音声信号と前記バイノーラルフラグ信号とを記録媒体に記録する記録部(14)と、

を備えて構成したことを特徴とする映像音声記録装置(101, 102, 103, 104, 105, 107)。

[2] 前記映像音声記録装置(101, 102, 103, 104, 105, 107)に内蔵されている内蔵マイク(21)と、

外部マイク接続端子(32)と、

前記外部マイク接続端子(32)に接続して使用する外部マイクとして、前記バイノーラルマイク(3)かバイノーラルマイク以外のマイクかを設定する設定部(48, 47)と、

前記外部マイク接続端子(32)に外部マイクが接続されているか否かを検出する接続検出部(41)と、

前記オーディオ処理部(22)に供給する音声信号として、前記内蔵マイク(21)より

出力された音声信号と前記外部マイクより出力された音声信号とを切り替えるスイッチ(Sw1)と、

前記設定部(47, 48)によって前記外部マイクがバイノーラルマイク(3)に設定され、前記接続検出部(41)によって前記外部マイク接続端子(32)に外部マイクが接続されていることが検出されたとき、音声モードを前記バイノーラルモードとする制御部(47)とを備え、

前記制御部(47)は、音声モードが前記バイノーラルモードのとき、前記スイッチ(Sw1)が前記外部マイクより出力された音声信号を前記オーディオ処理部(22)に供給するよう前記スイッチ(Sw1)を切り替え制御し、前記フラグ情報生成部(42)が前記バイノーラルフラグ信号を生成するよう前記フラグ情報生成部(42)を制御することを特徴とする請求の範囲第1項に記載の映像音声記録装置(101, 102, 103, 104, 105, 107)。

[3] 前記カメラ部(11)より出力された映像信号を表示する表示部(17)を備え、

音声モードが前記バイノーラルモードのとき、前記表示部(17)に前記バイノーラルモードであることを示すバイノーラルマークを表示させる表示制御部(47)を備えたことを特徴とする請求の範囲第1項に記載の映像音声記録装置(101, 102, 103, 104, 105, 107)。

[4] 前記カメラ部(11)は前記被写体を拡大して撮像するズーム機能を有し、

前記バイノーラルマイク(3)によって集音された音声信号を前記カメラ部(11)における拡大率に応じて増幅する音声ズーム処理部(33)を備えたことを特徴とする請求の範囲第1項に記載の映像音声記録装置(102)。

[5] 前記カメラ部(101)は前記被写体を拡大して撮像するズーム機能を有し、

前記バイノーラルマイク(3)が集音する音声信号の音源を仮想的に受聴者に接近させて仮想音源を形成するための頭部伝達関数を前記仮想音源と前記受聴者との複数の距離に対応させて記憶する伝達関数メモリ(334)と、この伝達関数メモリ(334)に記憶された複数の頭部伝達関数のいずれかを前記カメラ部(11)における拡大率に応じて選択的に取得する関数取得器(333)と、前記バイノーラルマイク(3)によって集音された音声信号に対して前記関数取得器(333)にて取得された頭部伝達

関数を畳み込む畳み込み演算器(336)とを有する音声ズーム処理部(33a, 33c)を備えたことを特徴とする請求の範囲第1項に記載の映像音声記録装置(102, 103)。

[6] 被写体を撮像した映像信号と、前記被写体から発せられた音を含む撮影者(300)の周囲音を集音した音声信号とを記録する映像音声記録方法において、

前記被写体を撮像する撮像ステップ(S155)と、

前記周囲音を集音するマイクとして、前記撮影者(300)の耳に装着するバイノーラルマイク(3)を使用するかバイノーラルマイク以外のマイクを使用するかを切り替える切り替えステップ(S151)と、

前記被写体を撮像した映像信号を処理するビデオ処理ステップ(S156)と、

前記周囲音を集音するマイクより出力された音声信号を処理するオーディオ処理ステップ(S156)と、

前記切り替えステップ(S151)にて前記周囲音を集音するマイクとして前記バイノーラルマイク(3)に切り替えた場合に、前記周囲音を集音する際の音声モードをバイノーラルモードとして、バイノーラルモードであることを示すバイノーラルフラグ信号を生成するフラグ情報生成ステップ(S158)と、

前記ビデオ処理ステップ(S156)にて処理された映像信号と前記オーディオ処理ステップ(S156)にて処理された音声信号と前記バイノーラルフラグ信号とを記録媒体(44)に記録する記録ステップ(S161)と、

を有することを特徴とする映像音声記録方法。

[7] 被写体を撮像した映像信号と、前記被写体から発せられた音を含む撮影者(300)の周囲音を集音した音声信号とが記録された記録媒体(44)を再生する映像音声再生装置(101, 102, 103, 104, 105, 106, 107)において、

前記記録媒体に記録された記録信号を再生する再生部(14)と、

前記再生部(14)によって再生された前記記録信号から前記映像信号と前記音声信号とを分離する分離化部(15)と、

前記分離化部(15)によって分離された前記映像信号を処理するビデオ処理部(16)と、

前記分離化部(15)によって分離された前記音声信号を処理するオーディオ処理部(26)と、

前記記録媒体(44)に、前記周囲音を集音するマイクとして前記撮影者(300)の耳に装着するバイノーラルマイクを使用したことを示すバイノーラルフラグ信号が記録されているとき、前記バイノーラルフラグ信号を取得するフラグ情報取得部(36)と、

前記フラグ情報取得部(36)が前記バイノーラルフラグ信号を取得したとき、前記オーディオ処理部(26)によって処理された前記音声信号をスピーカ(53, 54)によって発音した場合に発生するクロストーク信号を打ち消すよう前記音声信号を処理するクロストークキャンセラ(27)とを備え、

前記クロストークキャンセラ(27)は前記音声信号に対して予め求めたフィルタ特性を畳み込むフィルタ(272a－272d)を有し、前記フィルタ特性は、測定信号を円筒型構造体に装着した一対のマイクで集音した音声信号を用いて計測した頭部伝達関数に基づいたフィルタ特性であることを特徴とする映像音声再生装置(101, 102, 103, 104, 105, 106, 107)。

[8] 被写体を撮像した映像信号と、前記被写体から発せられた音を含む撮影者(300)の周囲音を集音した音声信号とが記録された記録媒体(44)を再生する映像音声再生方法において、

前記記録媒体(44)に記録された記録信号を再生する再生ステップ(S181)と、

前記再生ステップにて再生された前記記録信号から前記映像信号と前記音声信号とを分離する分離化ステップ(S183)と、

前記分離化ステップ(S183)にて分離された前記映像信号を処理するビデオ処理ステップ(S184)と、

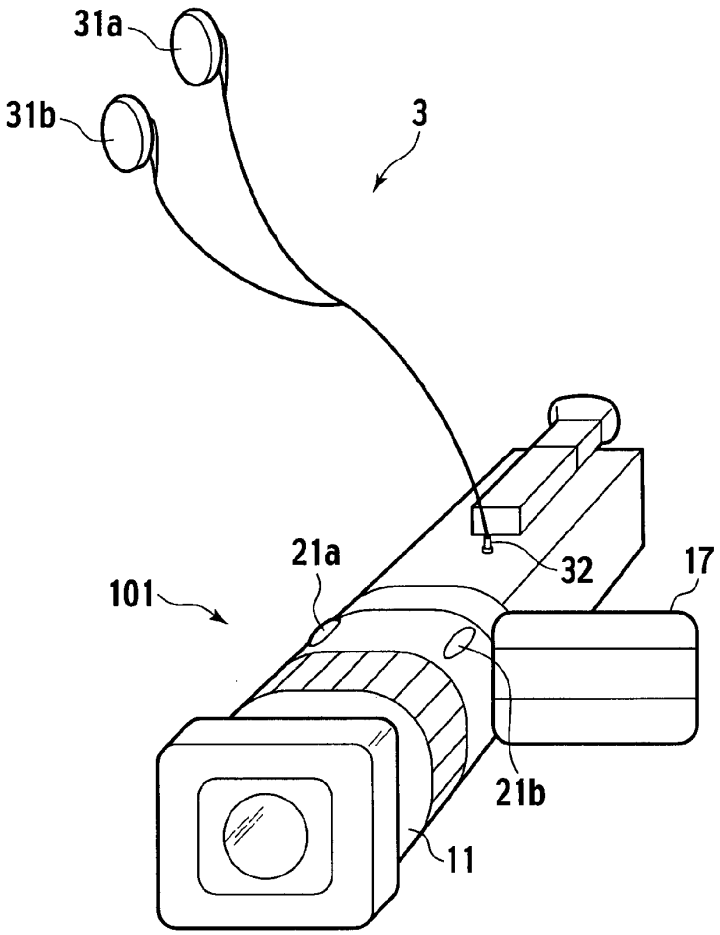
前記分離化ステップ(S183)によって分離された前記音声信号を処理するオーディオ処理ステップ(S184)と、

前記記録媒体(44)に、前記周囲音を集音するマイクとして前記撮影者(300)の耳に装着するバイノーラルマイク(3)を使用したことを示すバイノーラルフラグ信号が記録されているとき、前記バイノーラルフラグ信号を取得するフラグ情報取得ステップ(S186)と、

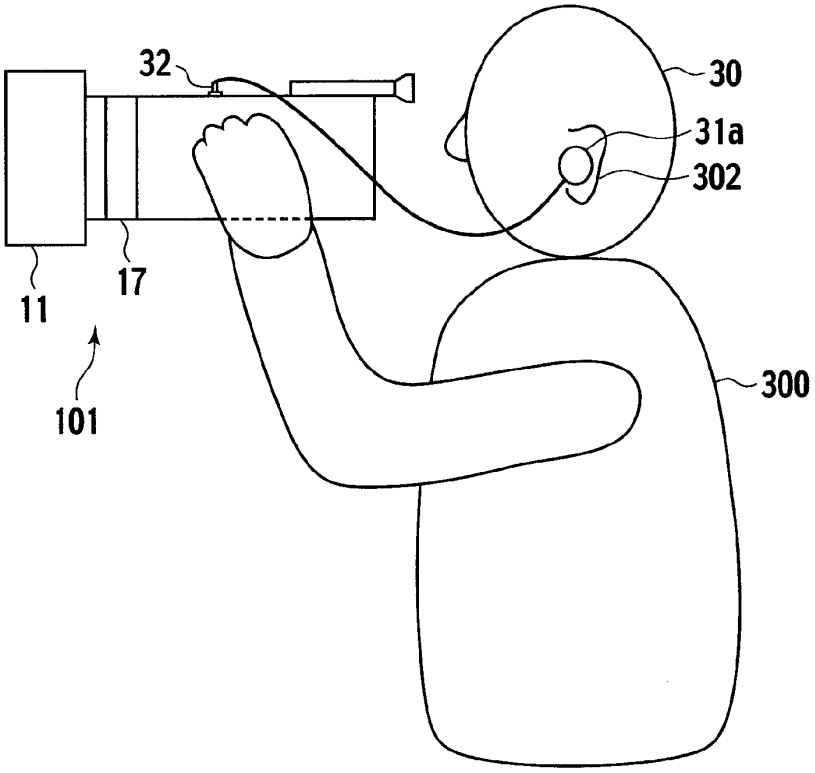
前記フラグ情報取得ステップ（S186）にて前記バイノーラルフラグ信号を取得したとき、前記オーディオ処理ステップ（S184）にて処理された前記音声信号をスピーカ（53，54）によって発音した場合に発生するクロストーク信号を打ち消すよう前記音声信号を処理するクロストークキャンセルステップ（S188）とを有し、

前記クロストークキャンセルステップ（S188）は前記音声信号に対して予め求めたフィルタ特性を畳み込むステップであり、前記フィルタ特性は、測定信号を円筒型構造体に装着した一対のマイクで集音した音声信号を用いて計測した頭部伝達関数に基づいたフィルタ特性であることを特徴とする映像音声再生方法。

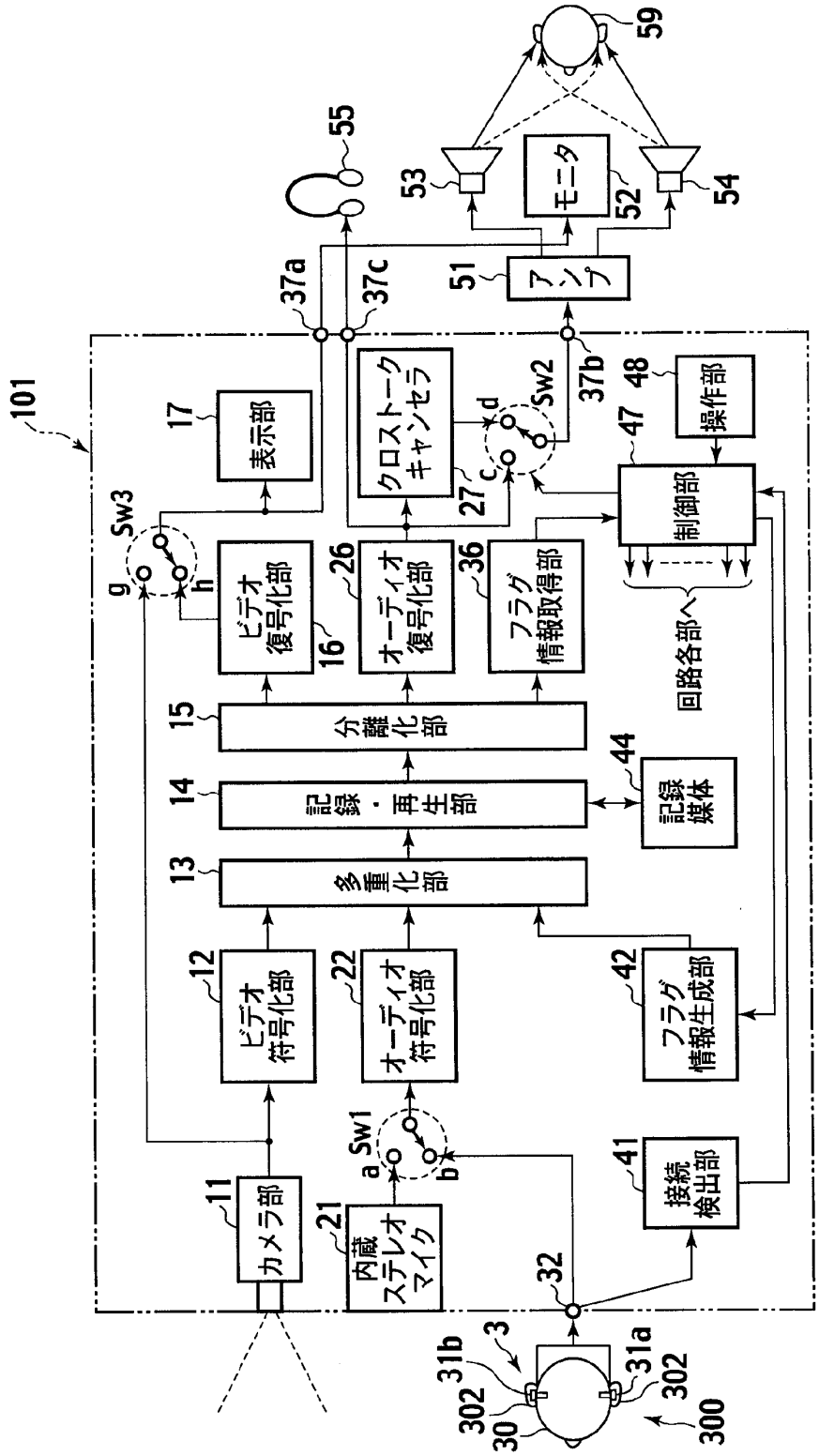
[図1]



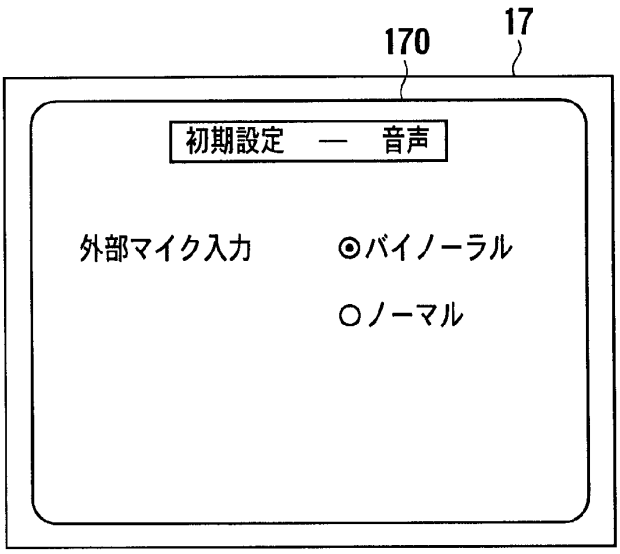
[図2]



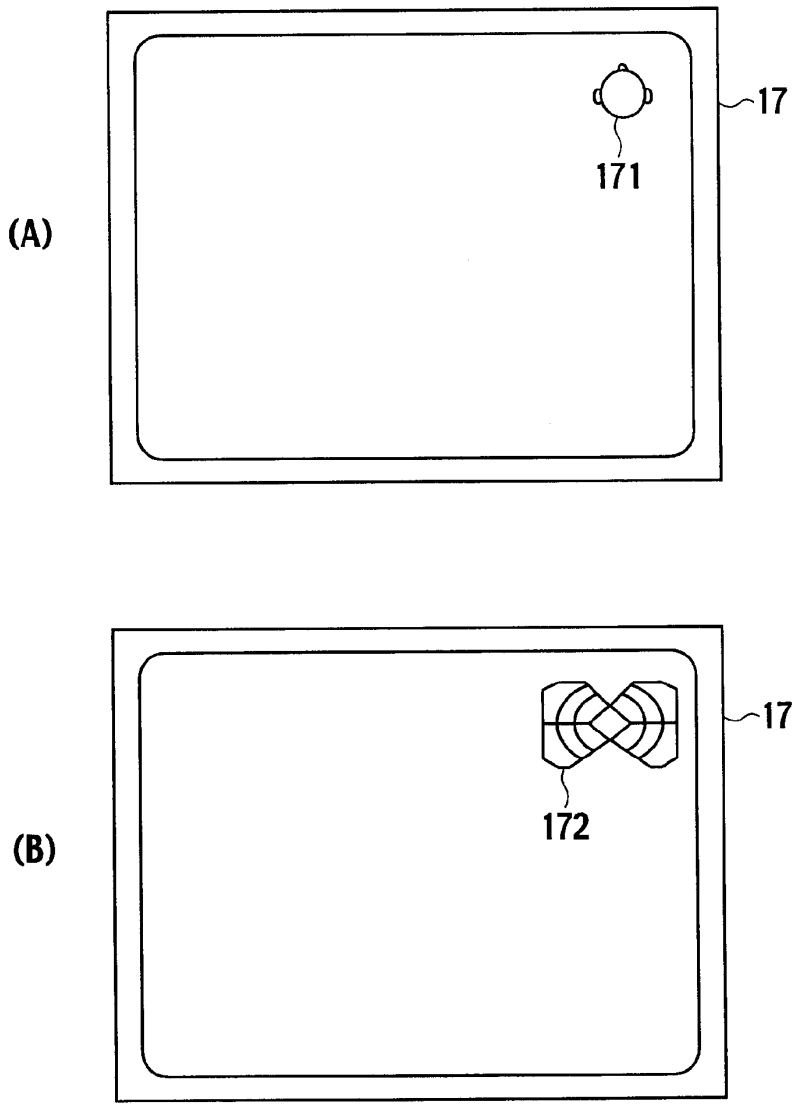
[図3]


101
カメラ部
11
ビデオ符号化部
12
内蔵ステレオマイク
21
オーディオ符号化部
22
Sw1
a
b
多重化部
13
記録・再生部
14
分離化部
15
ビデオ復号化部
16
オーディオ復号化部
26
フラグ情報取得部
36
表示部
17
Sw3
g
h
クロストークキャンセラ
27
Sw2
c
d
37a
37c
37b
アンプ
51
モニタ
52
53
54
55
59
操作部
48
制御部
47
回路各部へ
記録媒体
44
フラグ情報生成部
42
接続検出部
41
32
3
31a
31b
302
30
300

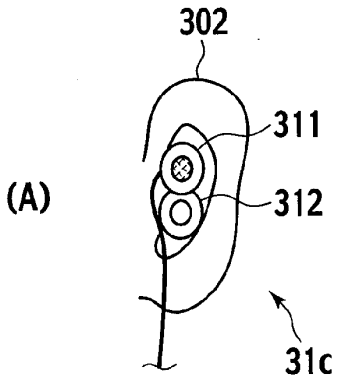
[図4]

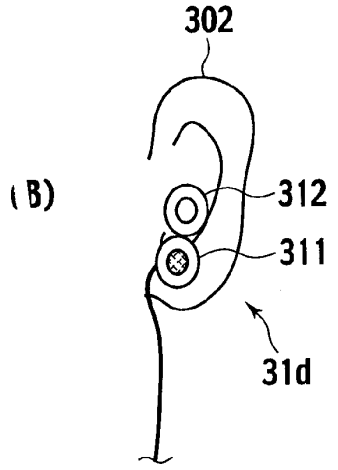


[図5]

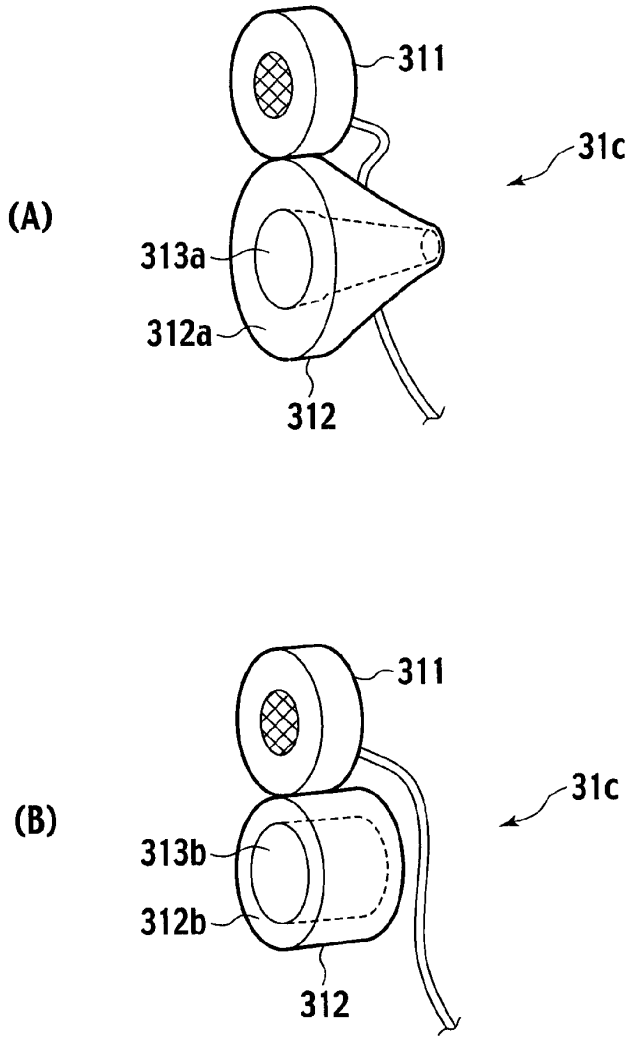


[図6]

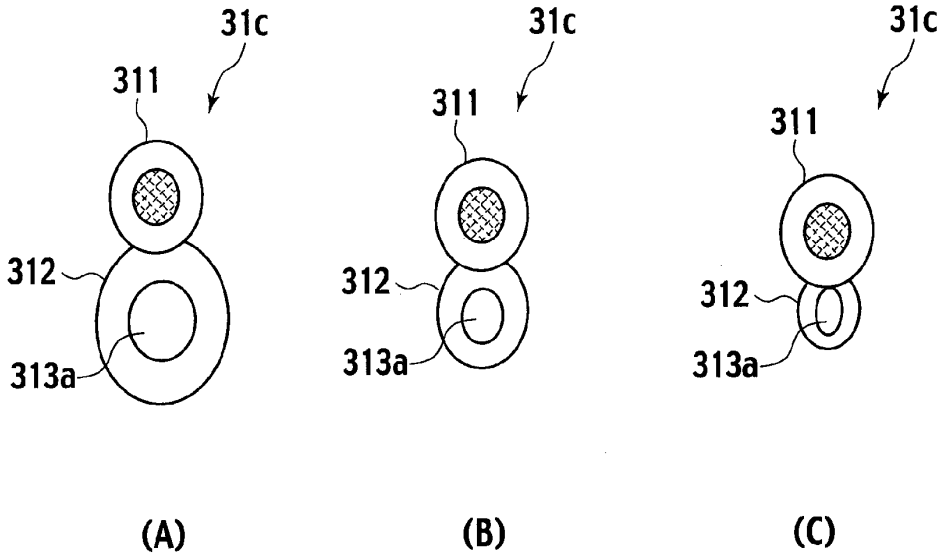




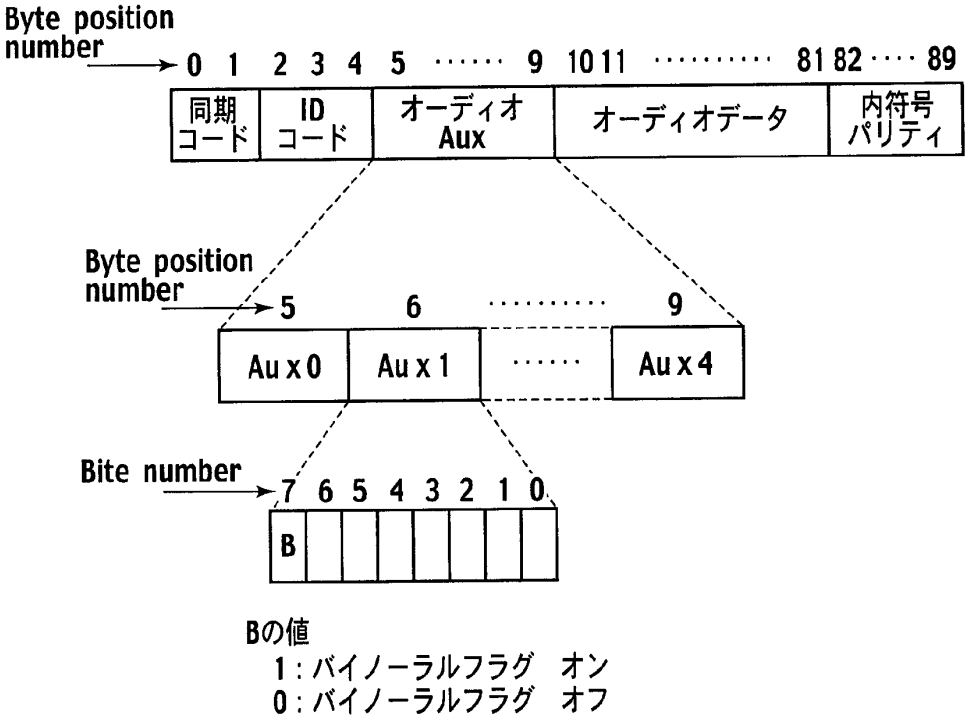
[図7]



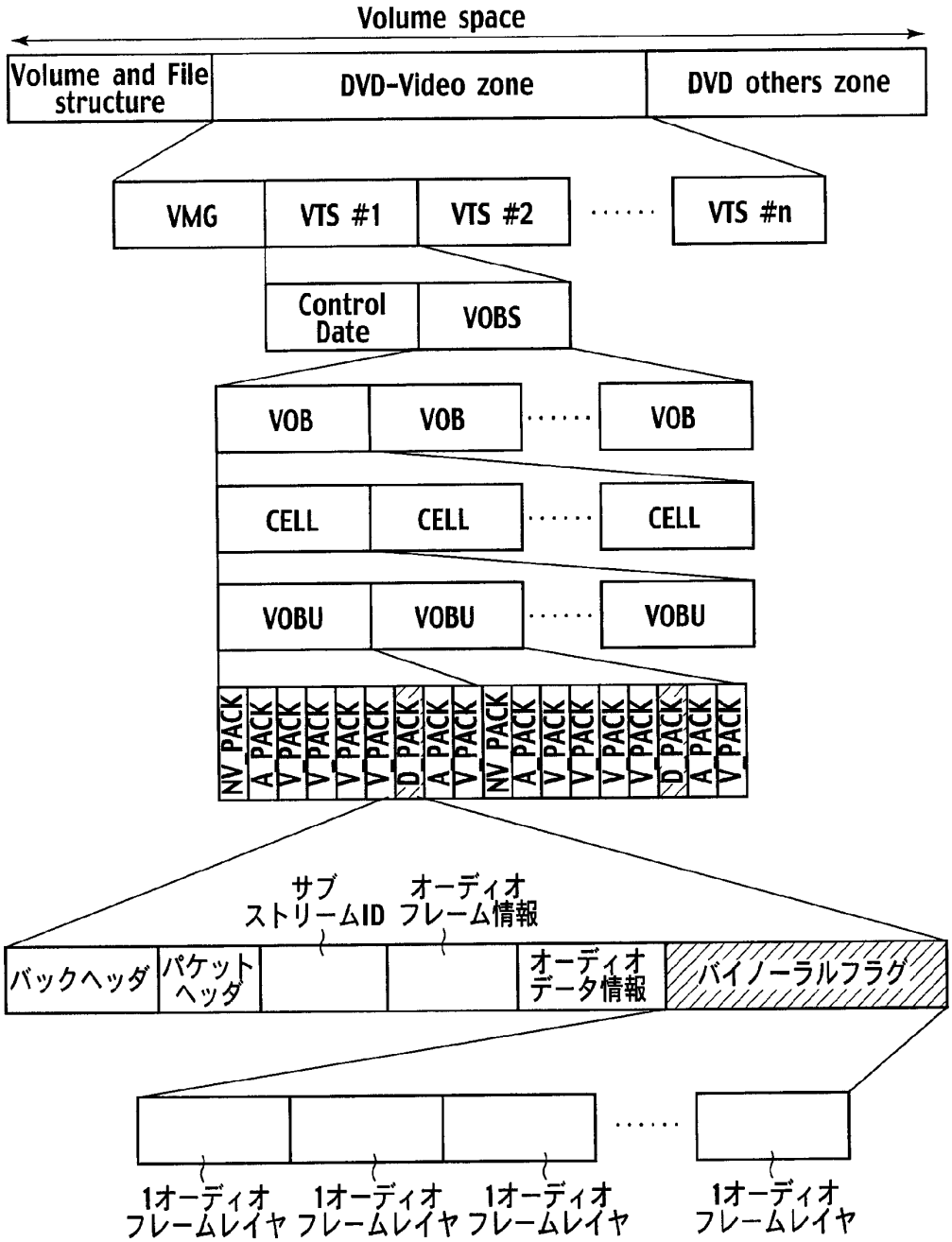
[図8]



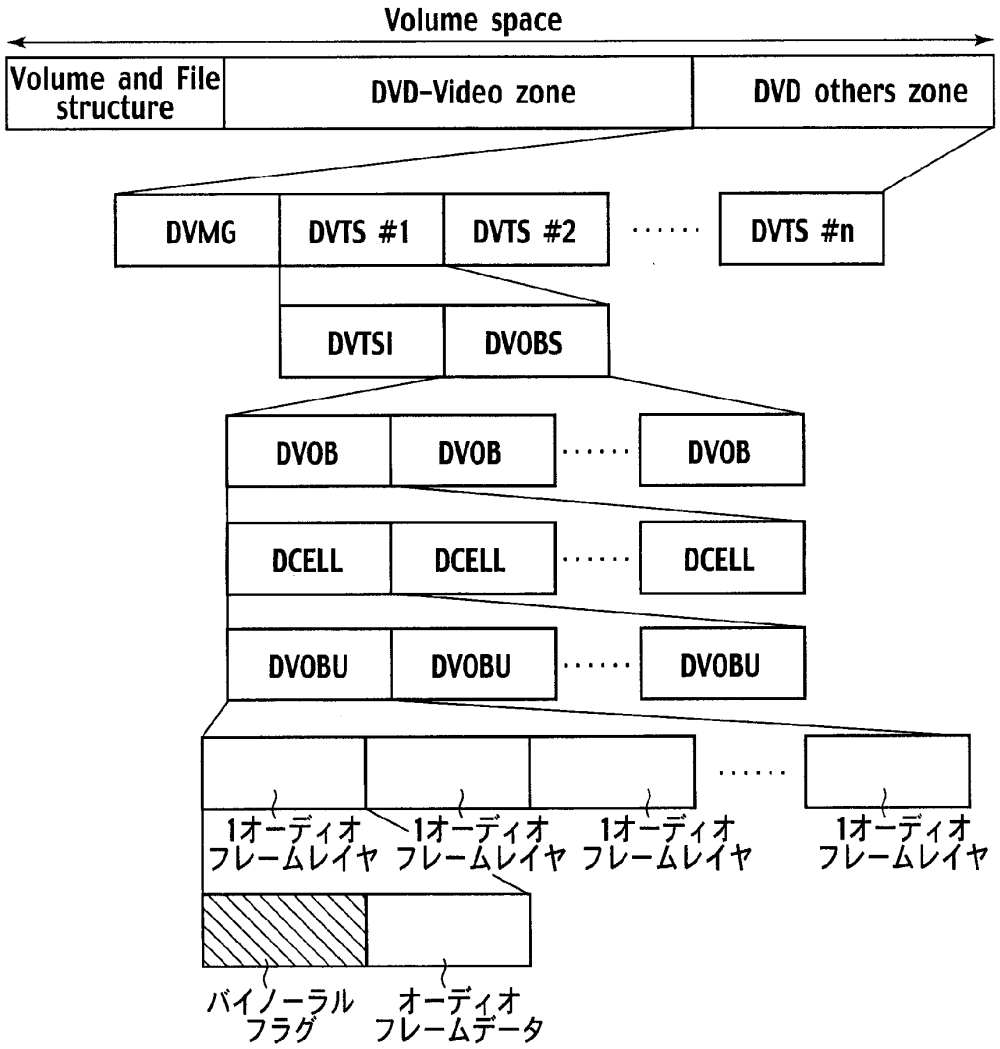
[図9]


Byte position number
0 1 2 3 4 5 ····· 9 10 11 ····· 81 82 ···· 89
同期コード
ID コード
オーディオ Aux
オーディオデータ
内符号パリティ
Byte position number
5 6 ····· 9
Au x 0
Au x 1
Au x 4
Bite number
7 6 5 4 3 2 1 0
B
Bの値
1：バイノーラルフラグ オン
0：バイノーラルフラグ オフ

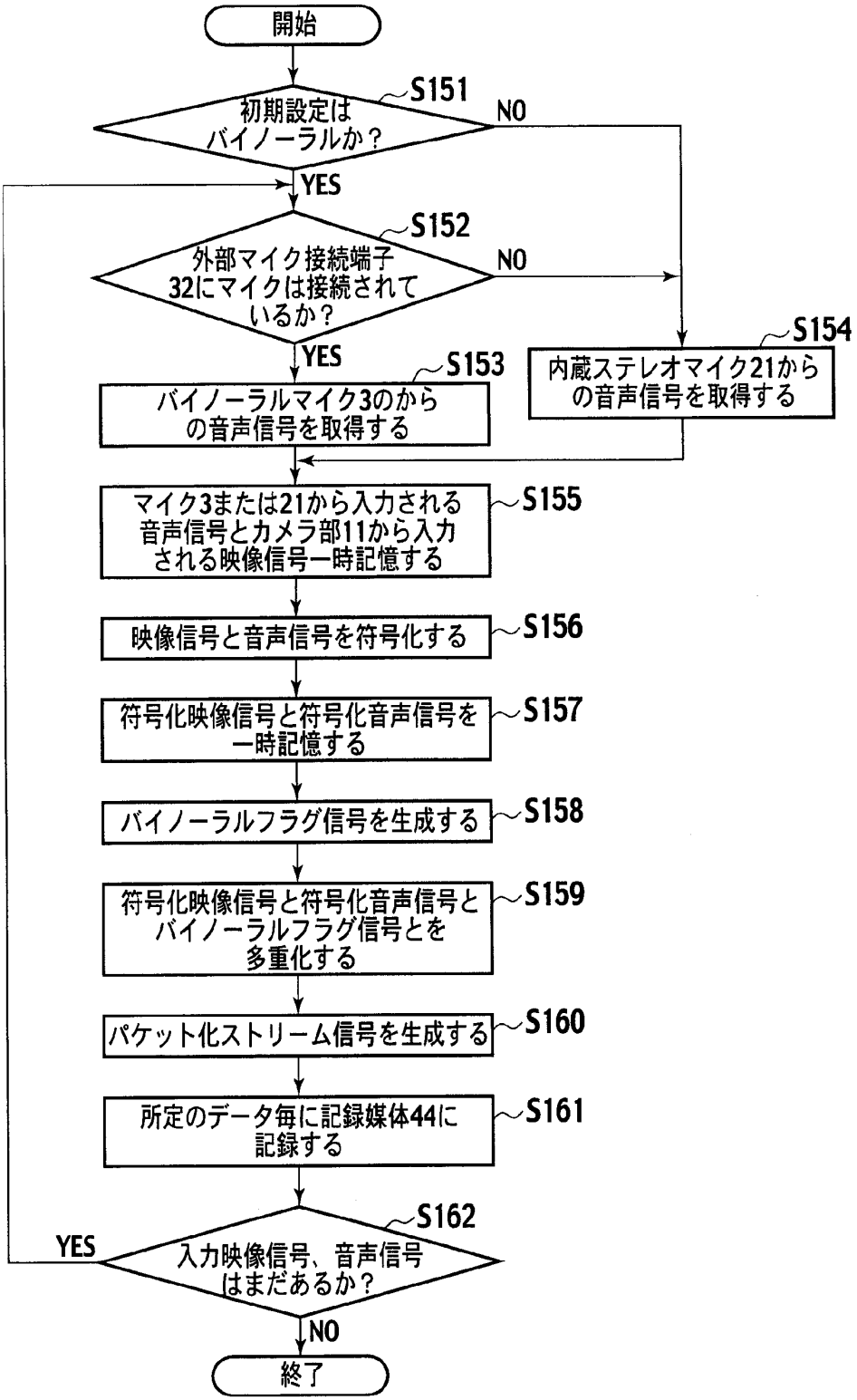
[図10]


Volume space
Volume and File structure
DVD-Video zone
DVD others zone
VMG
VTS #1
VTS #2
VTS #n
Control Date
VOBS
VOB
CELL
VOBU
NV PACK
A PACK
V PACK
D PACK
サブストリームID
オーディオフレーム情報
バックヘッダ
パケットヘッダ
オーディオデータ情報
バイノーラルフラグ
1オーディオフレームレイヤ

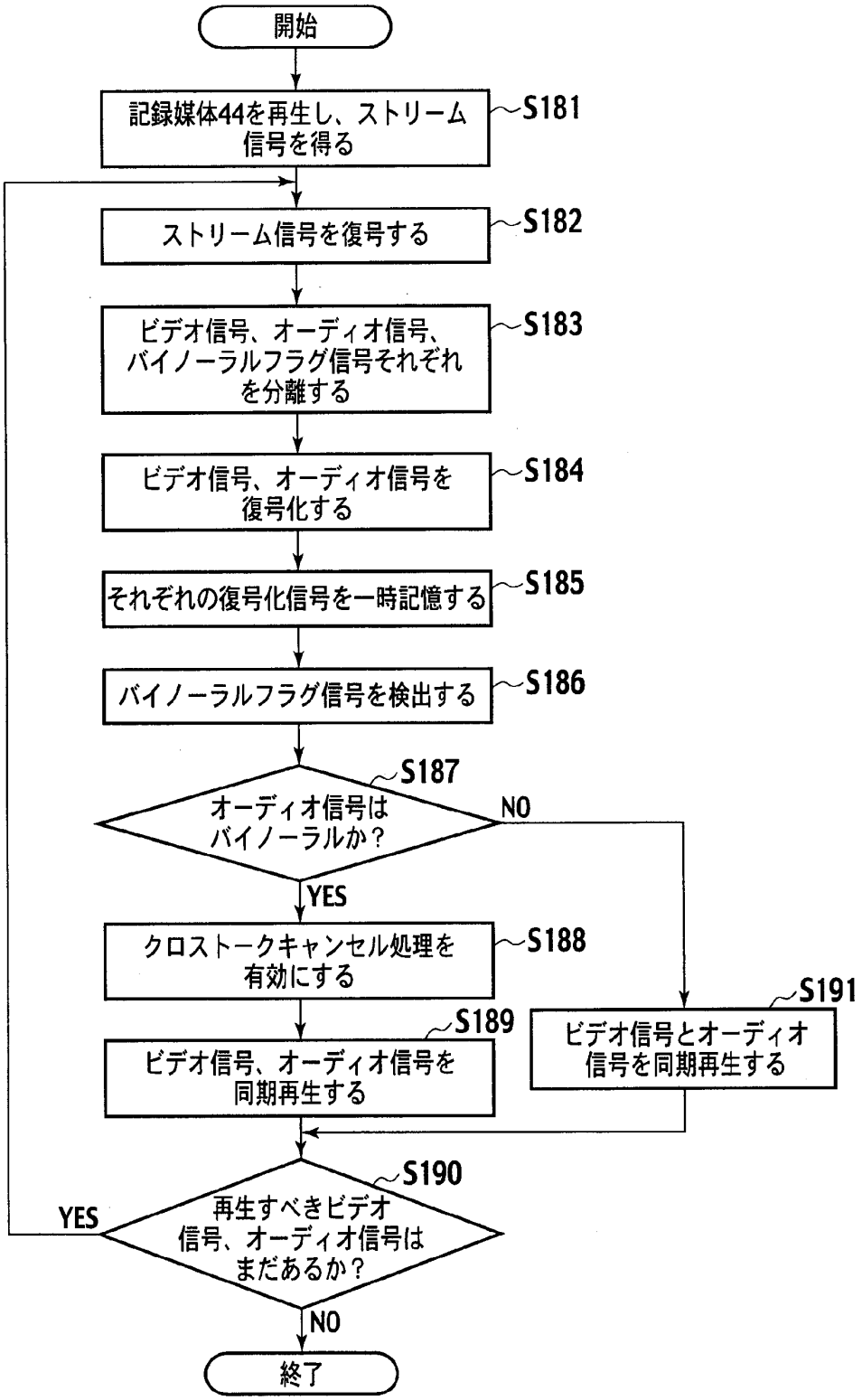
[図11]


Volume space
Volume and File structure
DVD-Video zone
DVD others zone
DVMG
DVTS #1
DVTS #2
DVTS #n
DVTSI
DVOBS
DVOB
DVOB
DVOB
DCELL
DCELL
DCELL
DVOBU
DVOBU
DVOBU
1オーディオフレームレイヤ
1オーディオフレームレイヤ
1オーディオフレームレイヤ
1オーディオフレームレイヤ
バイノーラルフラグ
オーディオフレームデータ

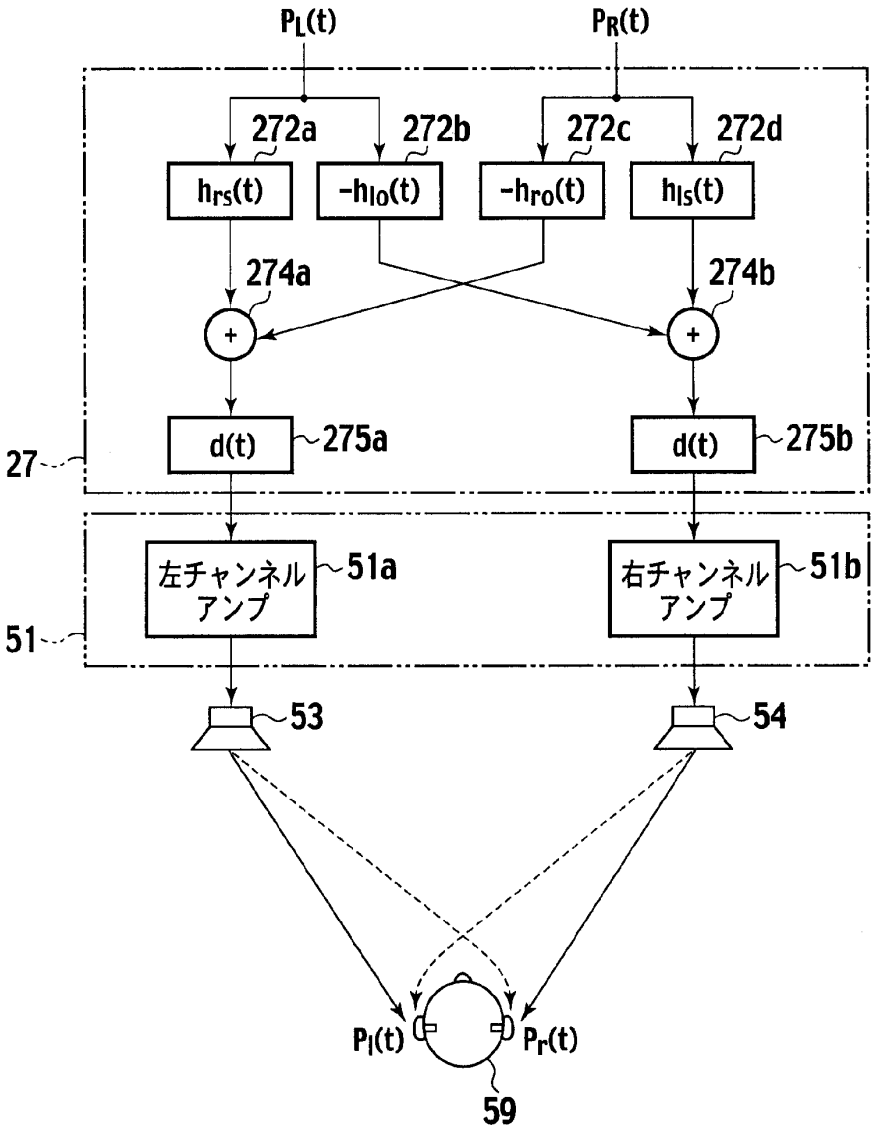
[図12]


開始
初期設定は
バイノーラルか？
S151
NO
YES
外部マイク接続端子
32にマイクは接続されて
いるか？
S152
NO
YES
バイノーラルマイク3のから
の音声信号を取得する
S153
内蔵ステレオマイク21から
の音声信号を取得する
S154
マイク3または21から入力される
音声信号とカメラ部11から入力
される映像信号一時記憶する
S155
映像信号と音声信号を符号化する
S156
符号化映像信号と符号化音声信号を
一時記憶する
S157
バイノーラルフラグ信号を生成する
S158
符号化映像信号と符号化音声信号と
バイノーラルフラグ信号とを
多重化する
S159
パケット化ストリーム信号を生成する
S160
所定のデータ毎に記録媒体44に
記録する
S161
入力映像信号、音声信号
はまだあるか？
S162
YES
NO
終了

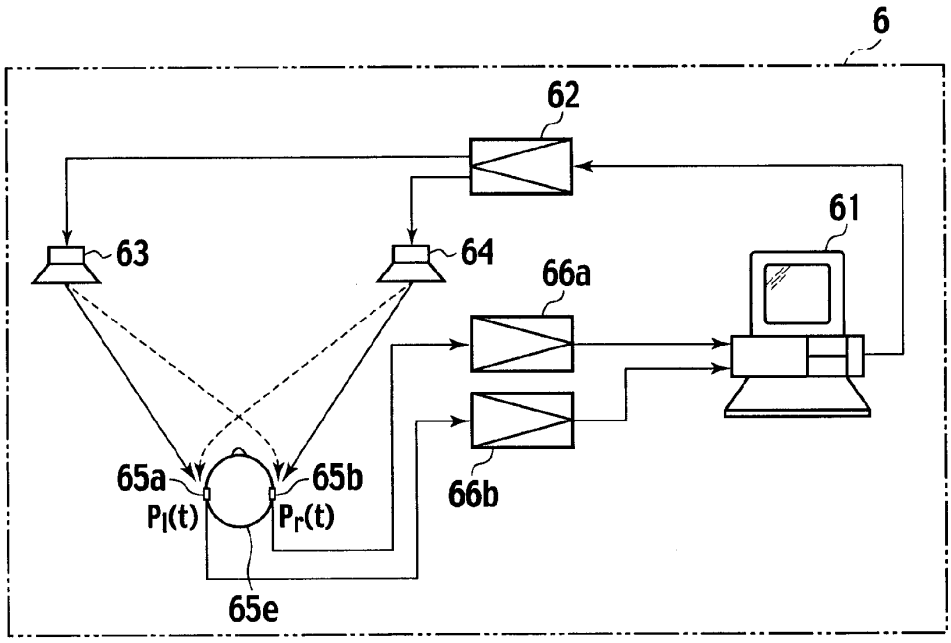
[図13]


開始
記録媒体44を再生し、ストリーム信号を得る
S181
ストリーム信号を復号する
S182
ビデオ信号、オーディオ信号、バイノーラルフラグ信号それぞれを分離する
S183
ビデオ信号、オーディオ信号を復号化する
S184
それぞれの復号化信号を一時記憶する
S185
バイノーラルフラグ信号を検出する
S186
S187
オーディオ信号はバイノーラルか？
NO
YES
クロストークキャンセル処理を有効にする
S188
S189
ビデオ信号、オーディオ信号を同期再生する
S191
ビデオ信号とオーディオ信号を同期再生する
S190
再生すべきビデオ信号、オーディオ信号はまだあるか？
YES
NO
終了

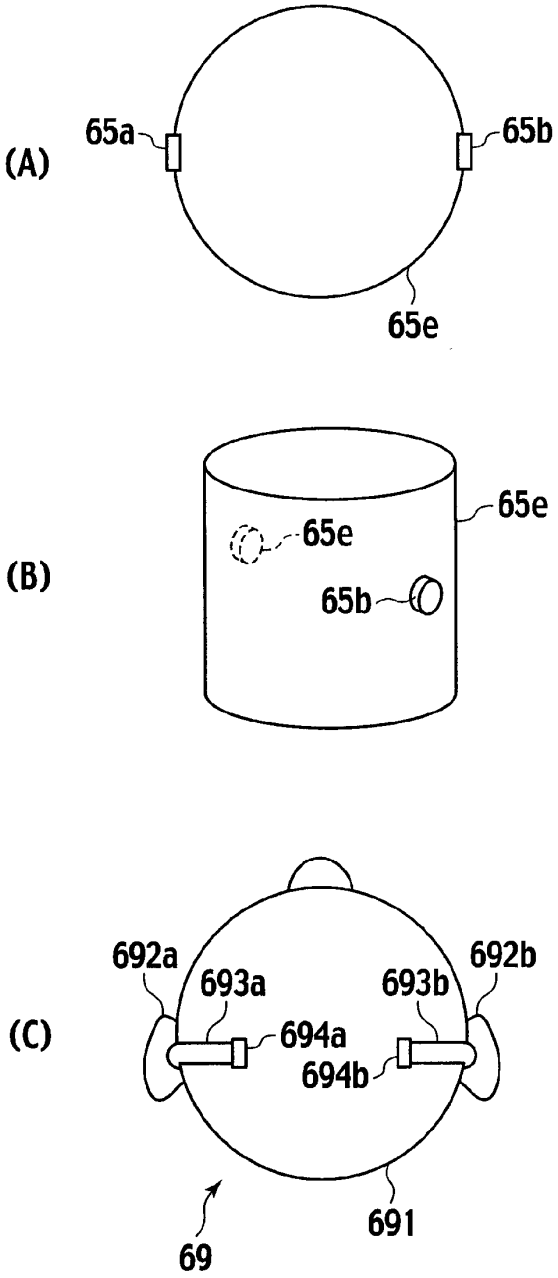
[図14]


PL(t)
PR(t)
272a
272b
272c
272d
hrs(t)
-hlo(t)
-hro(t)
hls(t)
274a
274b
+
+
d(t)
275a
d(t)
275b
27
左チャンネル
アンプ
51a
右チャンネル
アンプ
51b
51
53
54
Pl(t)
Pr(t)
59

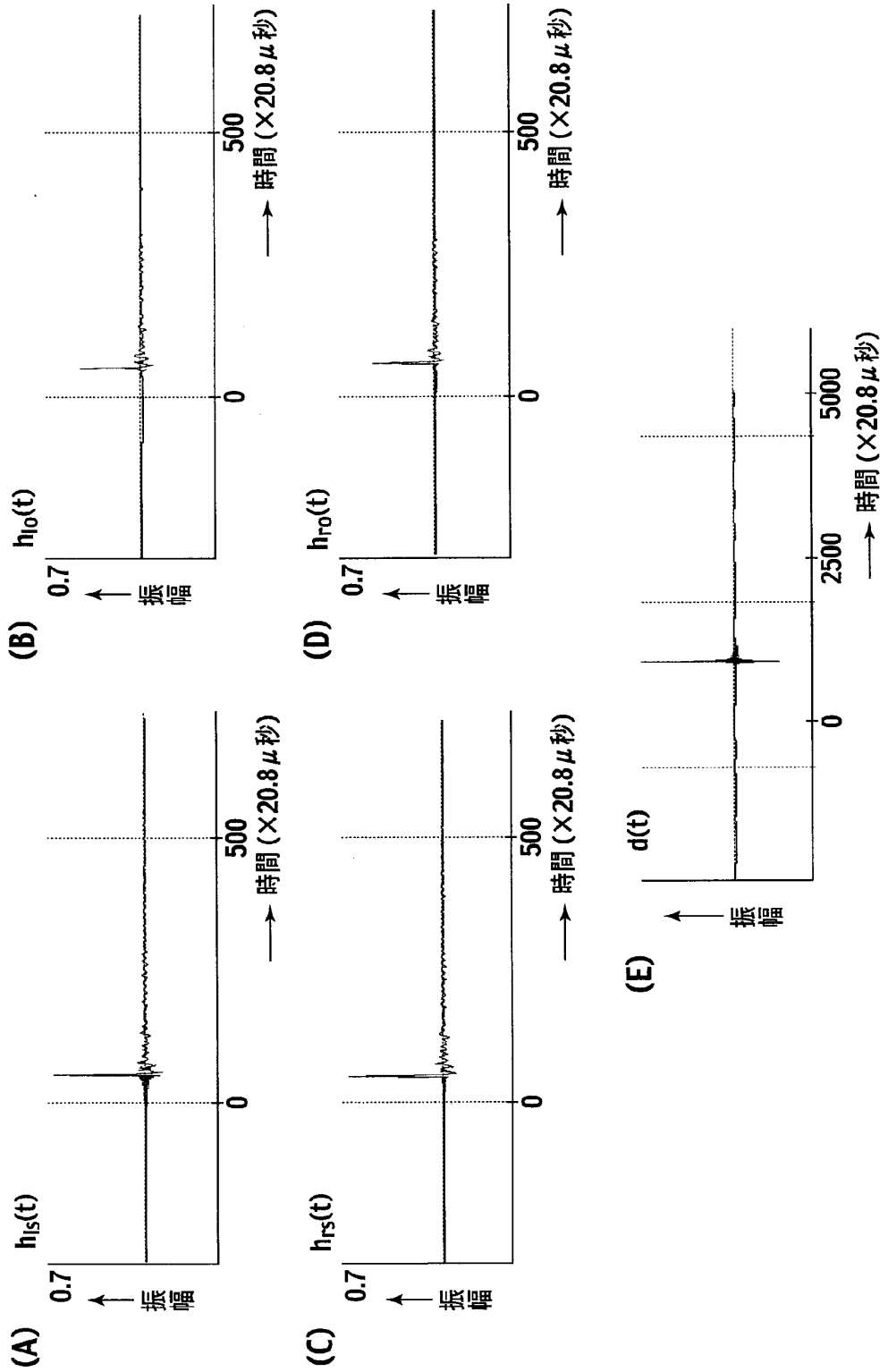
[図15]



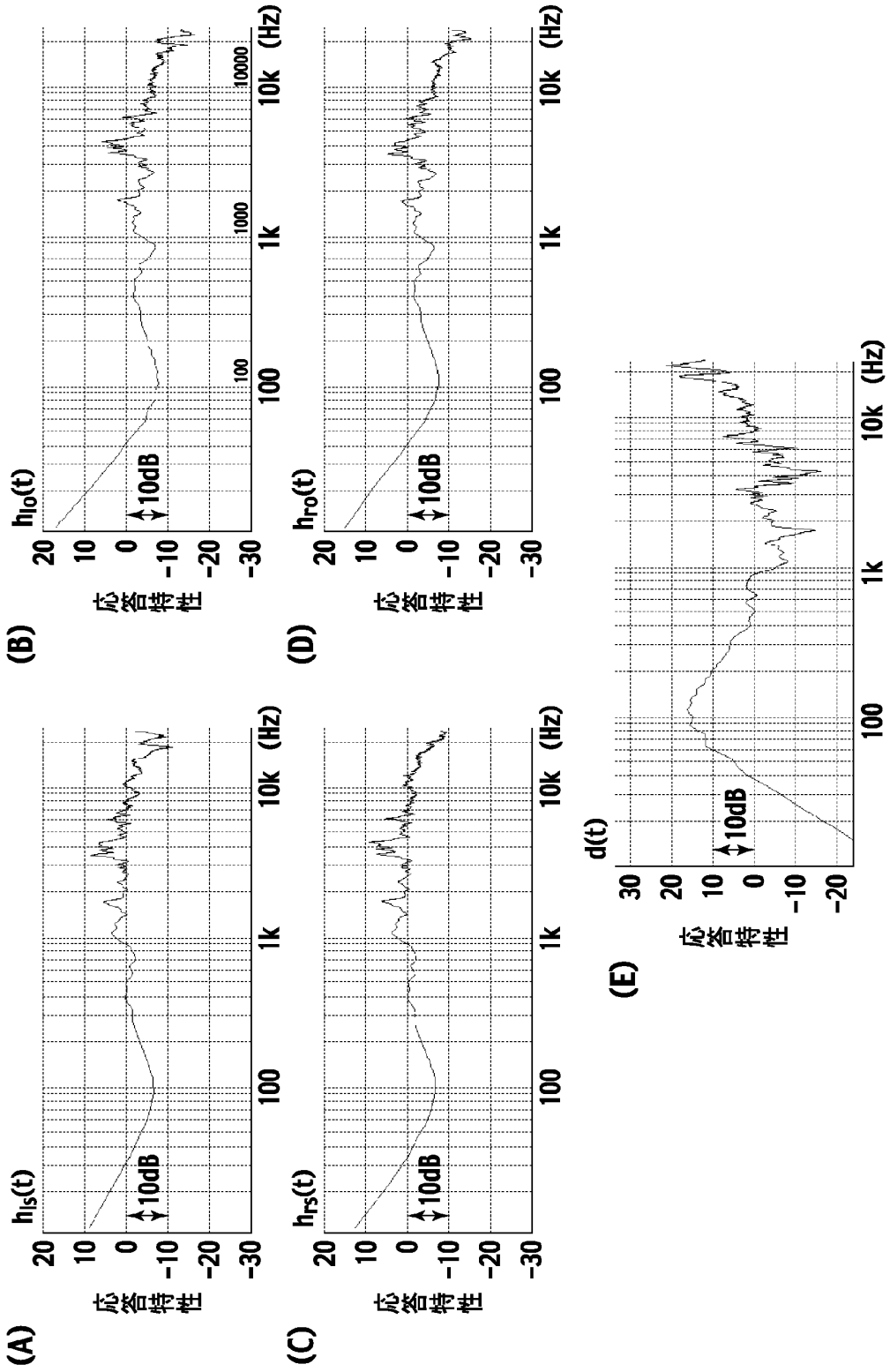
[図16]


(A)
65a
65b
65e
(B)
65e
65e
65b
(C)
692a
693a
693b
692b
694a
694b
691
69

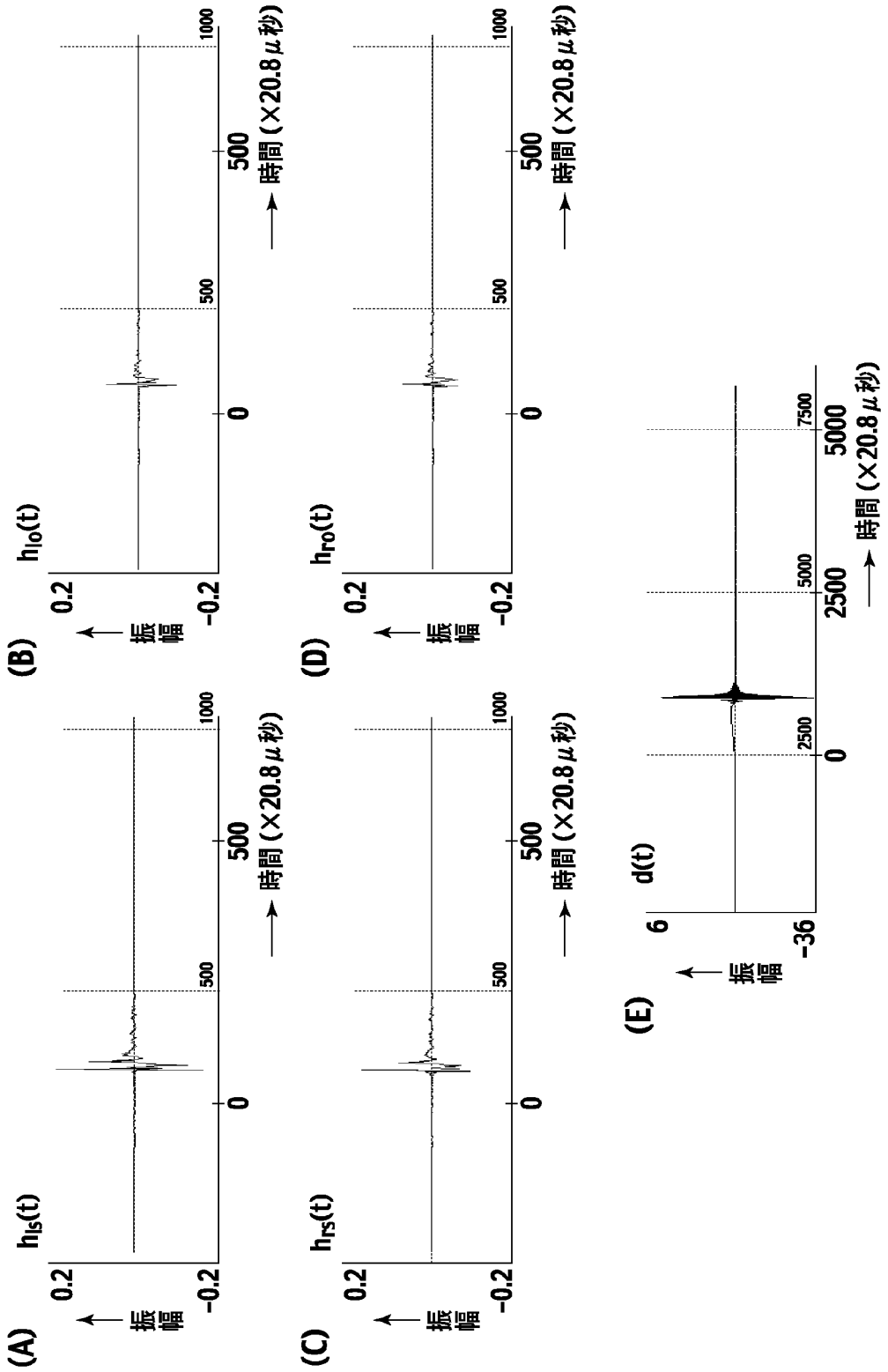
[図17]


(A)
$h_{IS}(t)$
0.7
振幅
0
500
時間 (×20.8μ秒)
(B)
$h_{IO}(t)$
0.7
振幅
0
500
時間 (×20.8μ秒)
(C)
$h_{rs}(t)$
0.7
振幅
0
500
時間 (×20.8μ秒)
(D)
$h_{ro}(t)$
0.7
振幅
0
500
時間 (×20.8μ秒)
(E)
d(t)
振幅
0
2500
5000
時間 (×20.8μ秒)

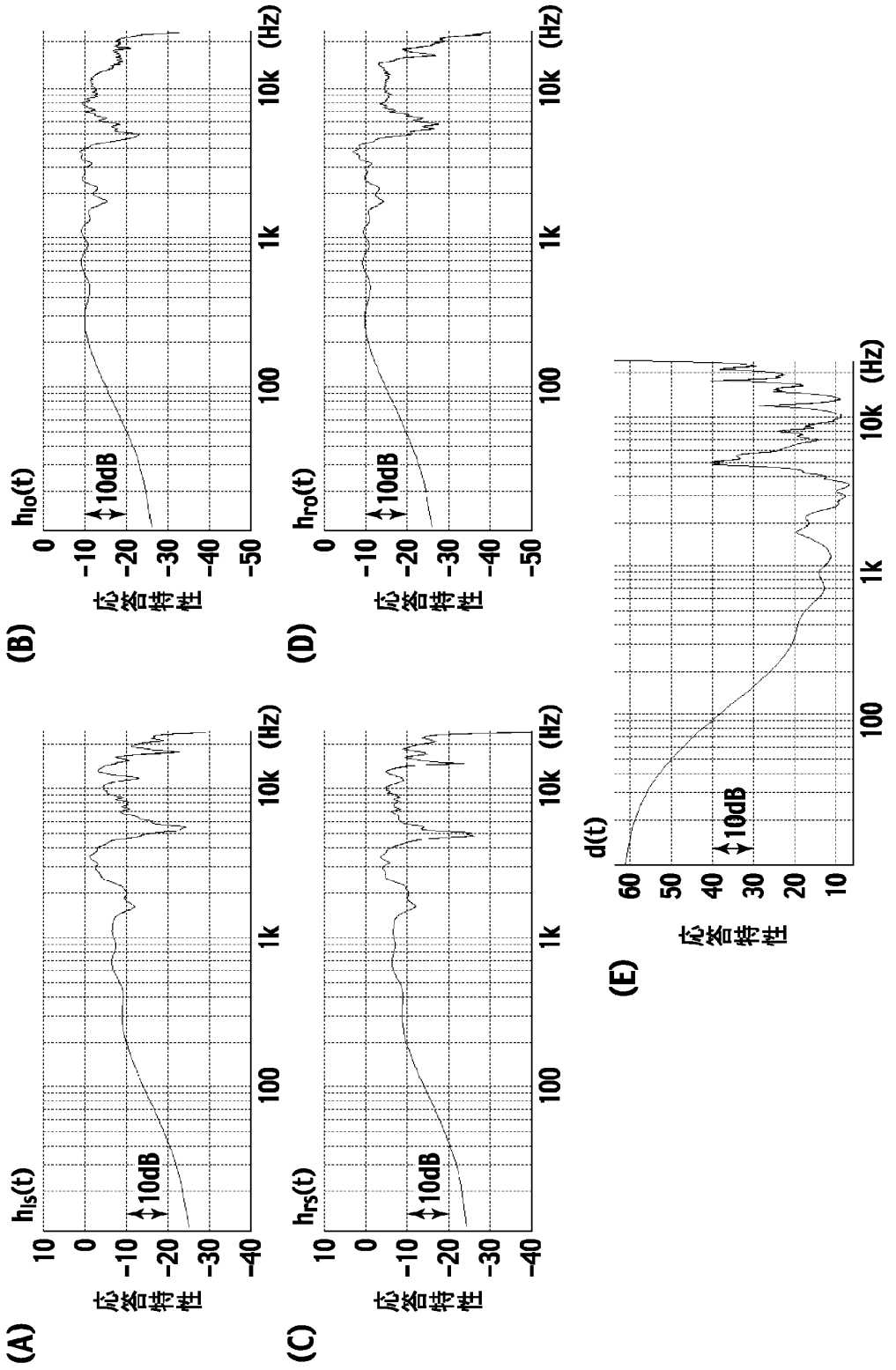
[図18]



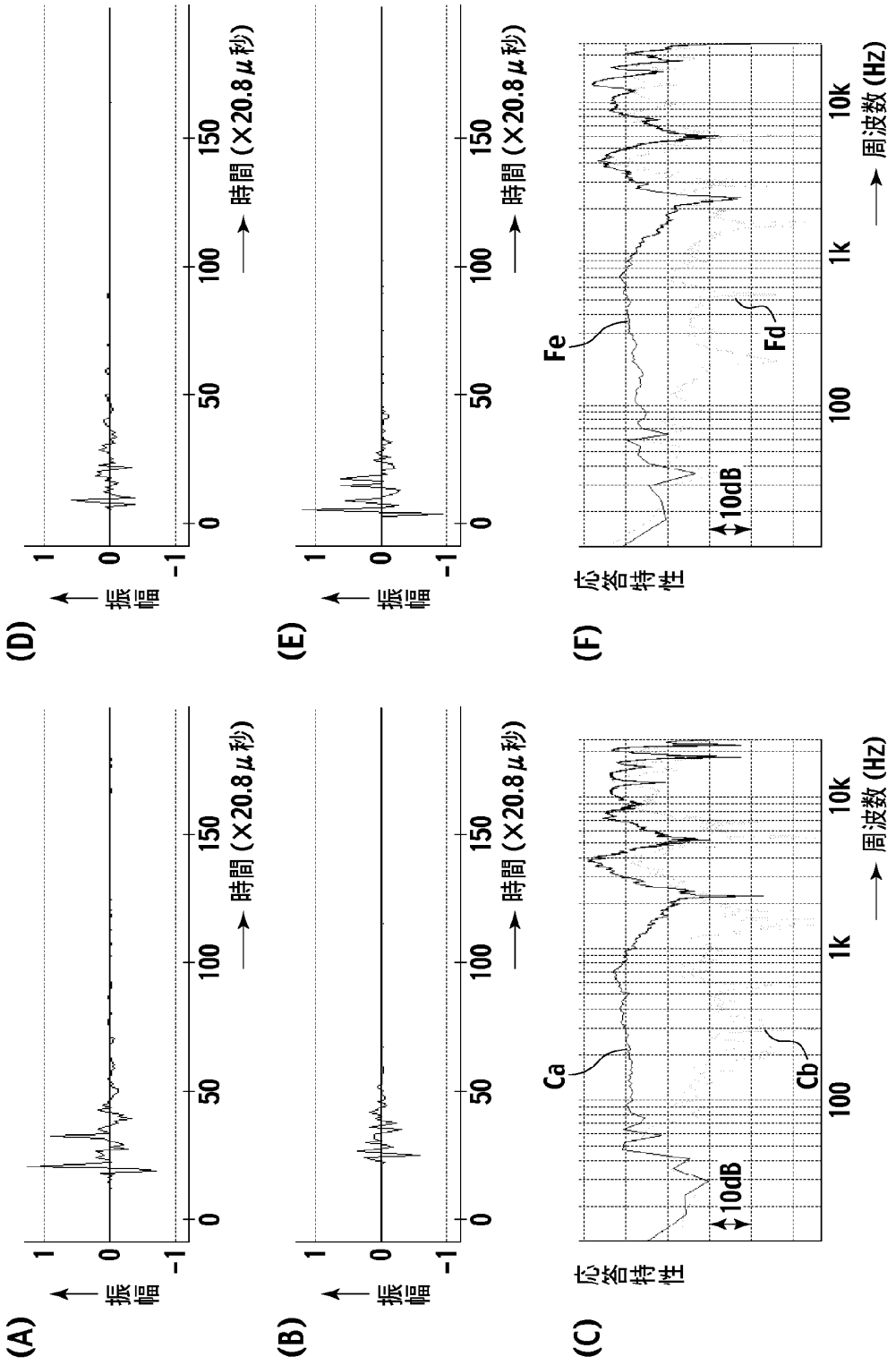
[図19]


(A)
$h_{ls}(t)$
0.2
-0.2
振幅
0
500
1000
時間 (×20.8μ秒)
(B)
$h_{lo}(t)$
0.2
-0.2
振幅
0
500
1000
時間 (×20.8μ秒)
(C)
$h_{rs}(t)$
0.2
-0.2
振幅
0
500
1000
時間 (×20.8μ秒)
(D)
$h_{ro}(t)$
0.2
-0.2
振幅
0
500
1000
時間 (×20.8μ秒)
(E)
d(t)
6
-36
振幅
0
2500
5000
7500
時間 (×20.8μ秒)

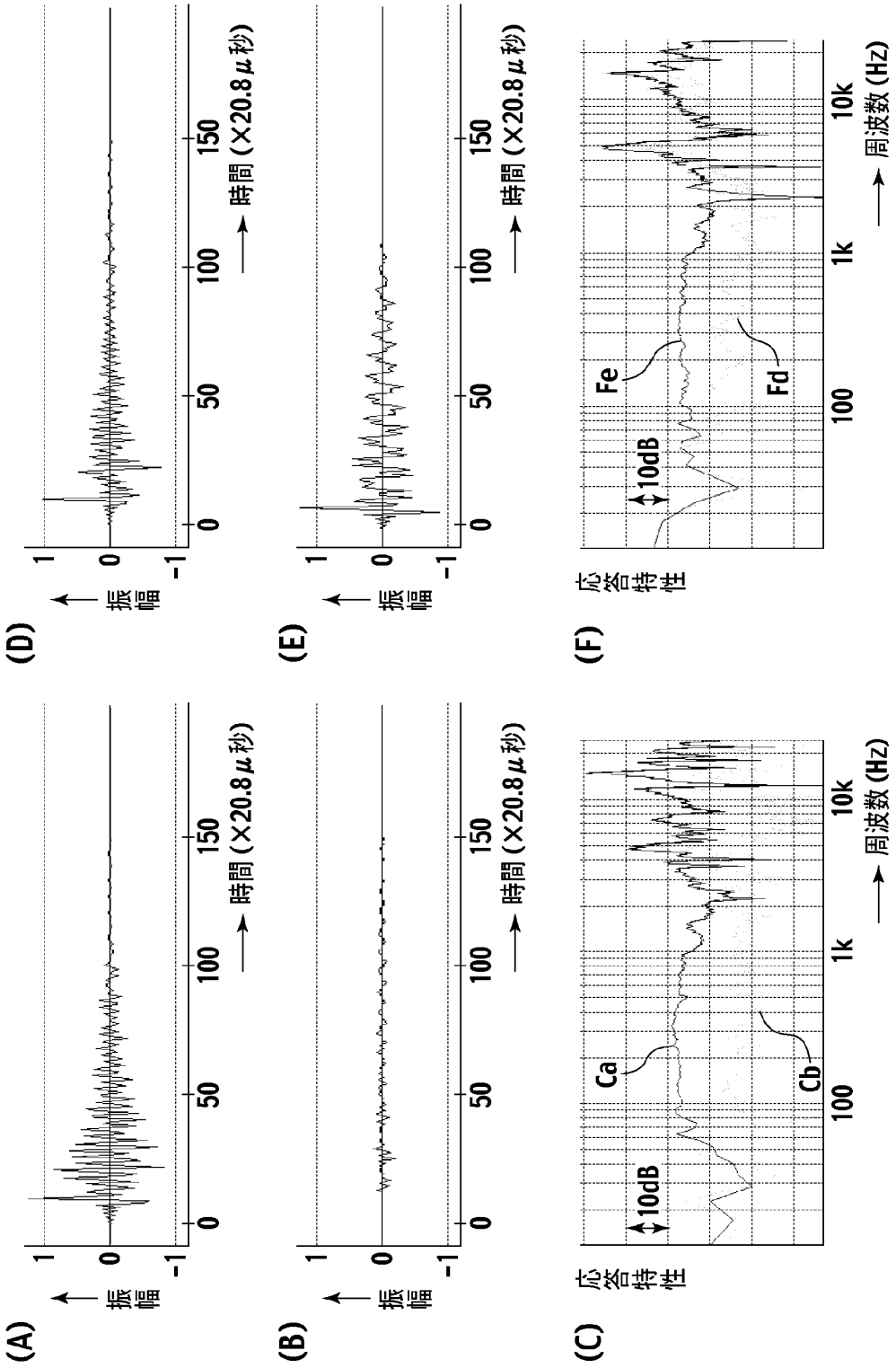
[図20]


(A)
$h_{ls}(t)$
10dB
10
0
−10
−20
−30
−40
伝達関数
100
1k
10k (Hz)
(B)
$h_{lo}(t)$
10dB
0
−10
−20
−30
−40
−50
伝達関数
100
1k
10k (Hz)
(C)
$h_{rs}(t)$
10dB
10
0
−10
−20
−30
−40
伝達関数
100
1k
10k (Hz)
(D)
$h_{ro}(t)$
10dB
0
−10
−20
−30
−40
−50
伝達関数
100
1k
10k (Hz)
(E)
d(t)
10dB
60
50
40
30
20
10
伝達関数
100
1k
10k (Hz)

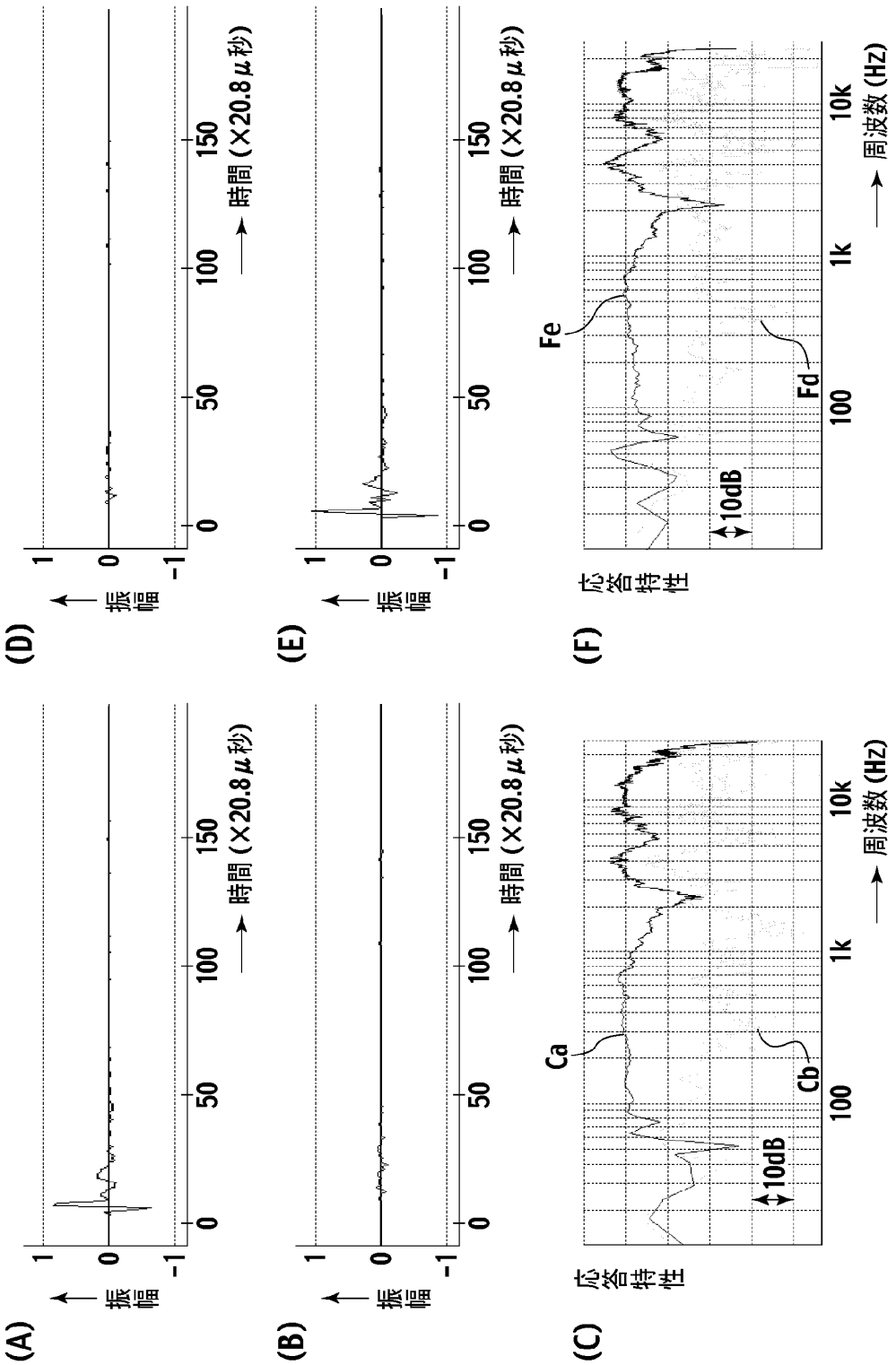
[図21]


(A)
(B)
(C)
(D)
(E)
(F)
振幅
時間 (×20.8μ秒)
周波数 (Hz)
応答特性
Ca
Cb
Fd
Fe
10dB
1
0
-1
50
100
150
1k
10k

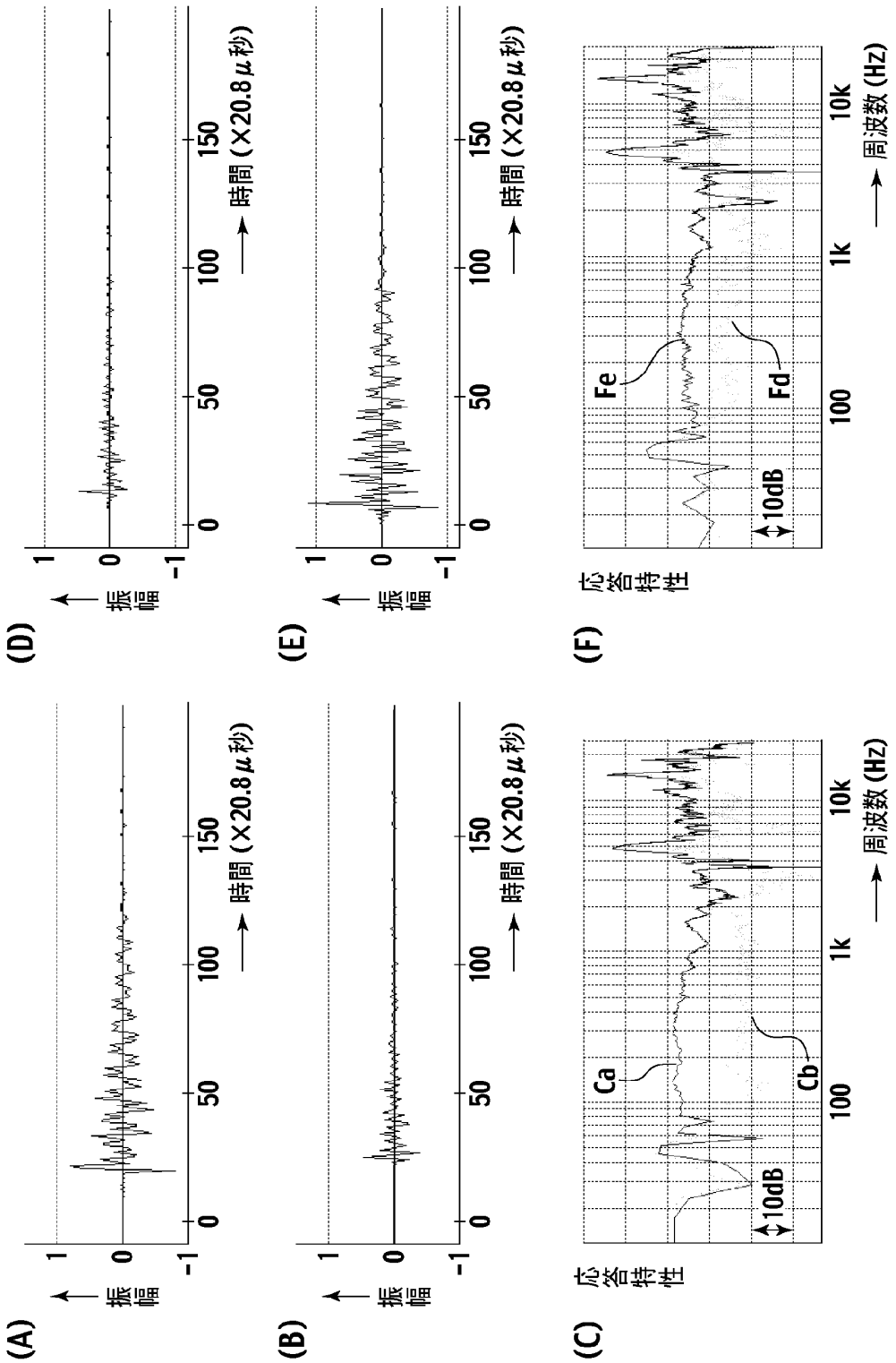
[図22]


(A)
(B)
(C)
(D)
(E)
(F)
振幅
時間 (×20.8μ秒)
周波数 (Hz)
応答特性
Ca
Cb
Fd
Fe
10dB
1
0
-1
50
100
150
1k
10k

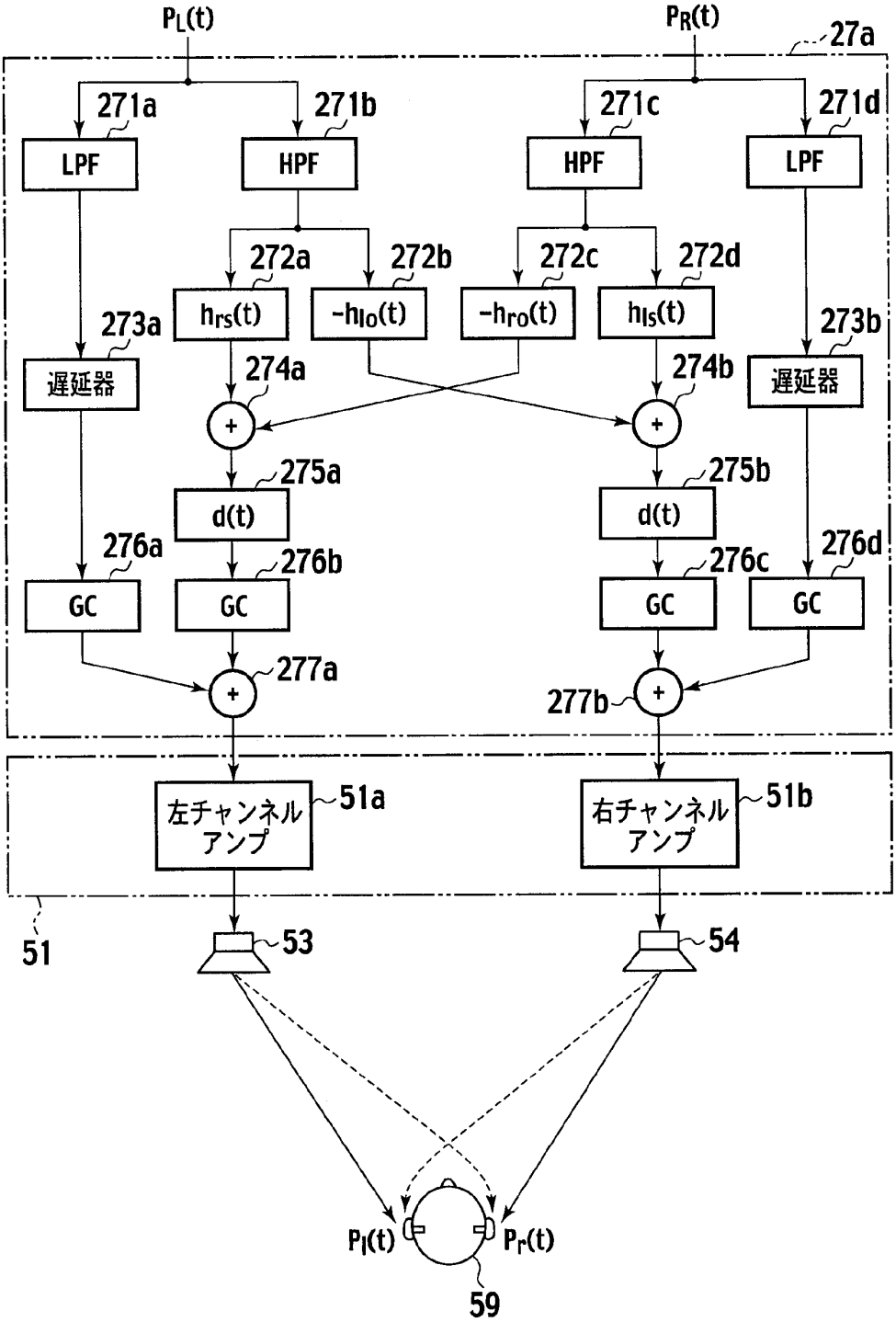
[図23]


(A)
(B)
(C)
(D)
(E)
(F)
振幅
時間 (×20.8μ秒)
周波数 (Hz)
応答特性
10dB
Ca
Cb
Fe
Fd
1
0
-1
50
100
150
1k
10k

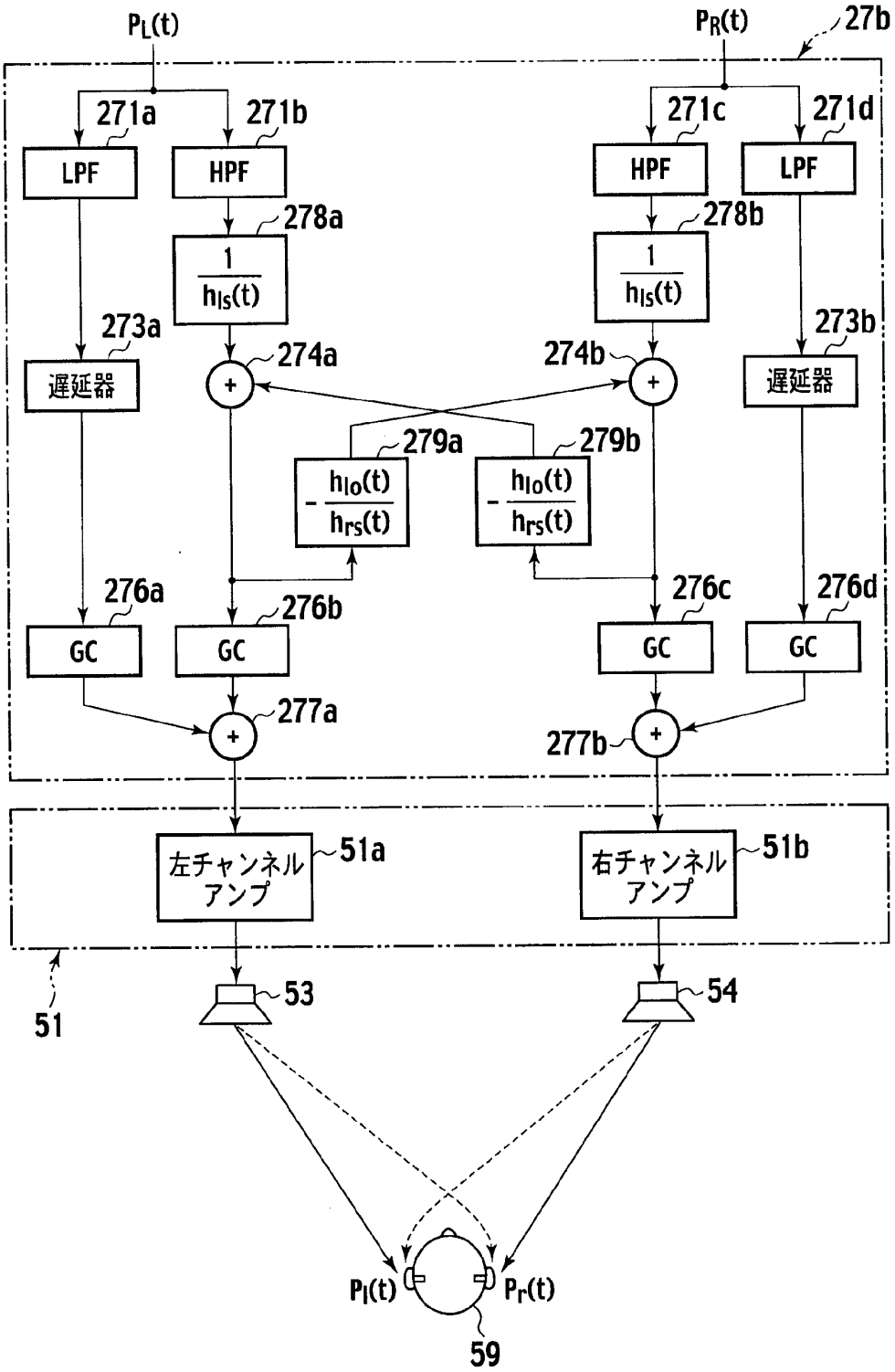
[図24]


(A)
(B)
(C)
(D)
(E)
(F)
振幅
時間 (×20.8μ秒)
周波数 (Hz)
応答特性
Ca
Cb
Fe
Fd
10dB
1
0
-1
50
100
150
1k
10k

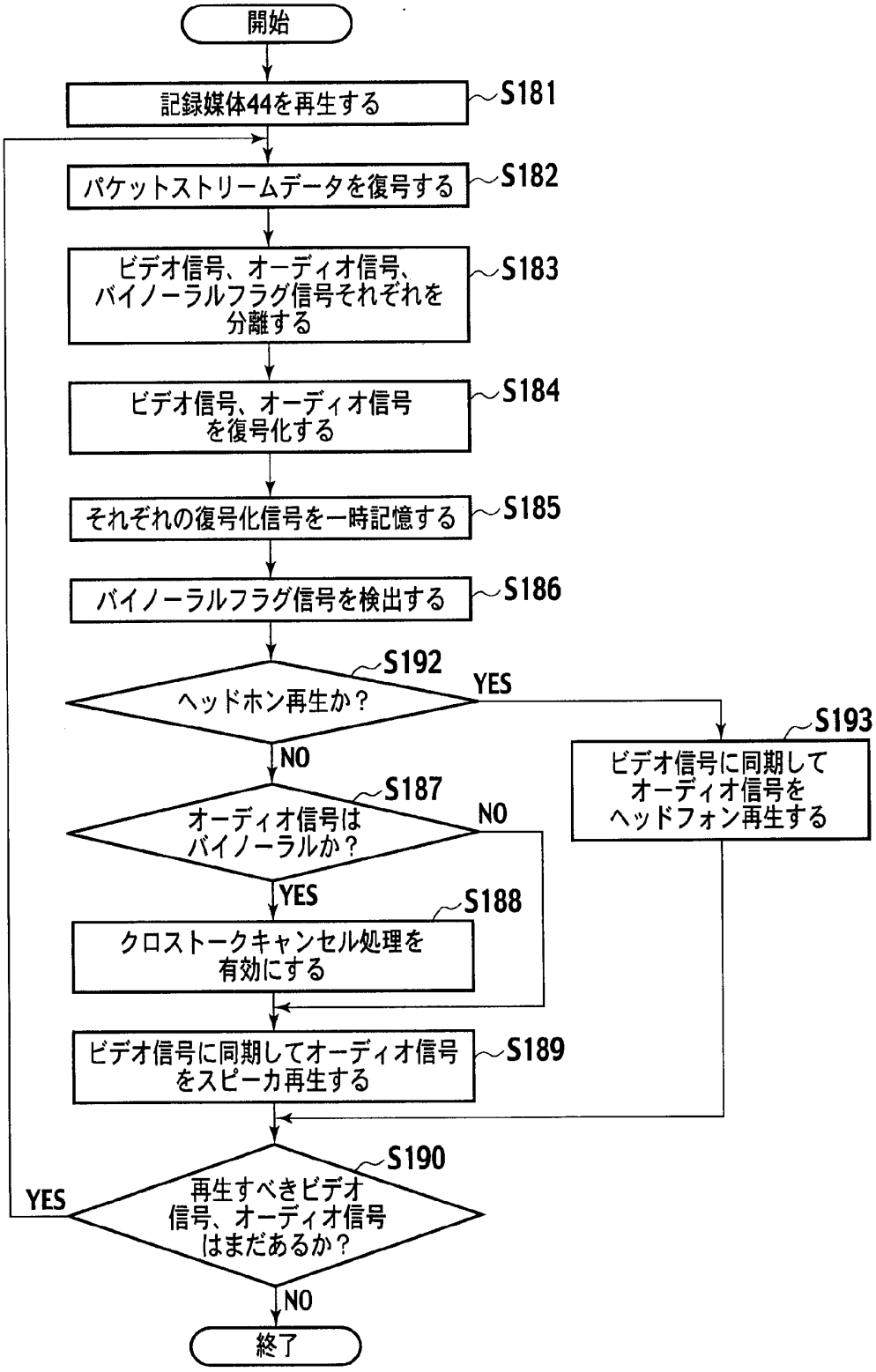
[図25]


PL(t)
PR(t)
27a
271a
271b
271c
271d
LPF
HPF
HPF
LPF
272a
272b
272c
272d
hrs(t)
-hlo(t)
-hro(t)
hls(t)
273a
273b
遅延器
遅延器
274a
274b
+
+
275a
275b
d(t)
d(t)
276a
276b
276c
276d
GC
GC
GC
GC
277a
277b
+
+
左チャンネル
アンプ
51a
右チャンネル
アンプ
51b
51
53
54
Pl(t)
Pr(t)
59

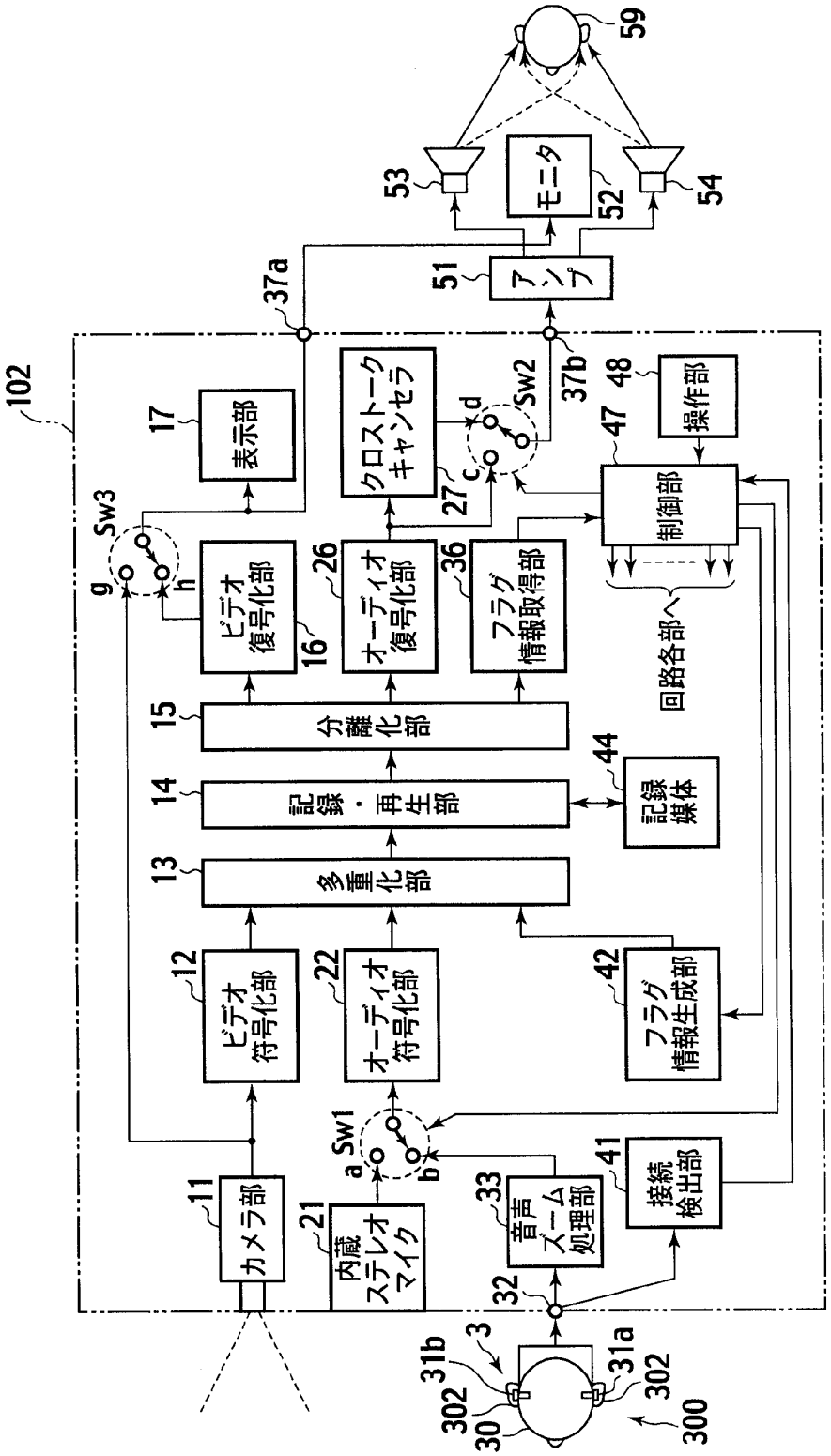
[図26]


PL(t)
PR(t)
27b
271a
271b
271c
271d
LPF
HPF
HPF
LPF
278a
278b
1/hIS(t)
1/hIS(t)
273a
273b
遅延器
遅延器
274a
274b
279a
279b
-hIO(t)/hRS(t)
-hIO(t)/hRS(t)
276a
276b
276c
276d
GC
GC
GC
GC
277a
277b
左チャンネルアンプ
51a
右チャンネルアンプ
51b
51
53
54
PI(t)
Pr(t)
59

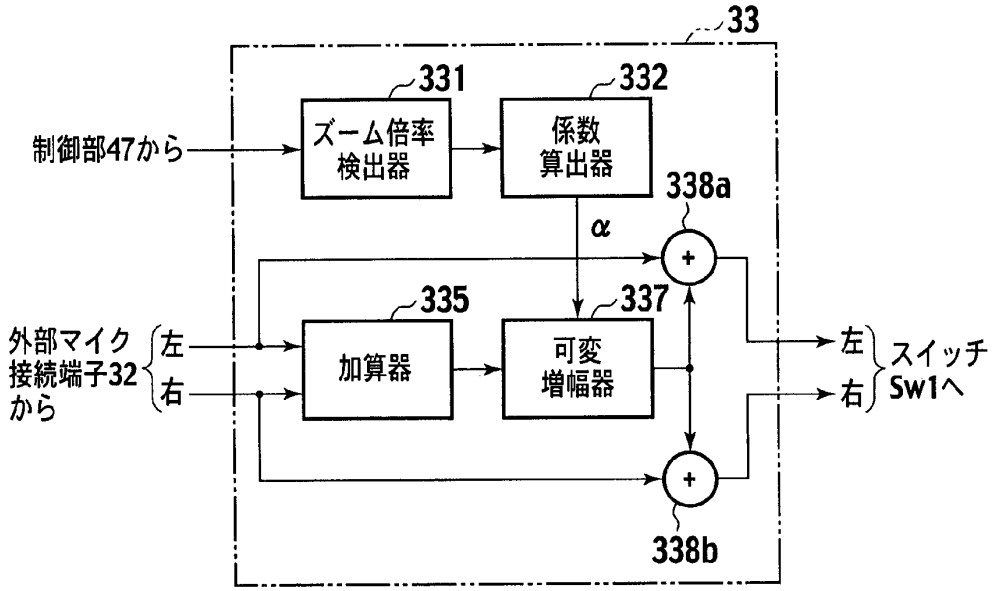
[図27]


開始
記録媒体44を再生する
S181
パケットストリームデータを復号する
S182
ビデオ信号、オーディオ信号、
バイノーラルフラグ信号それぞれを
分離する
S183
ビデオ信号、オーディオ信号
を復号化する
S184
それぞれの復号化信号を一時記憶する
S185
バイノーラルフラグ信号を検出する
S186
S192
ヘッドホン再生か？
YES
S193
ビデオ信号に同期して
オーディオ信号を
ヘッドフォン再生する
NO
S187
オーディオ信号は
バイノーラルか？
NO
YES
S188
クロストークキャンセル処理を
有効にする
ビデオ信号に同期してオーディオ信号
をスピーカ再生する
S189
S190
再生すべきビデオ
信号、オーディオ信号
はまだあるか？
YES
NO
終了

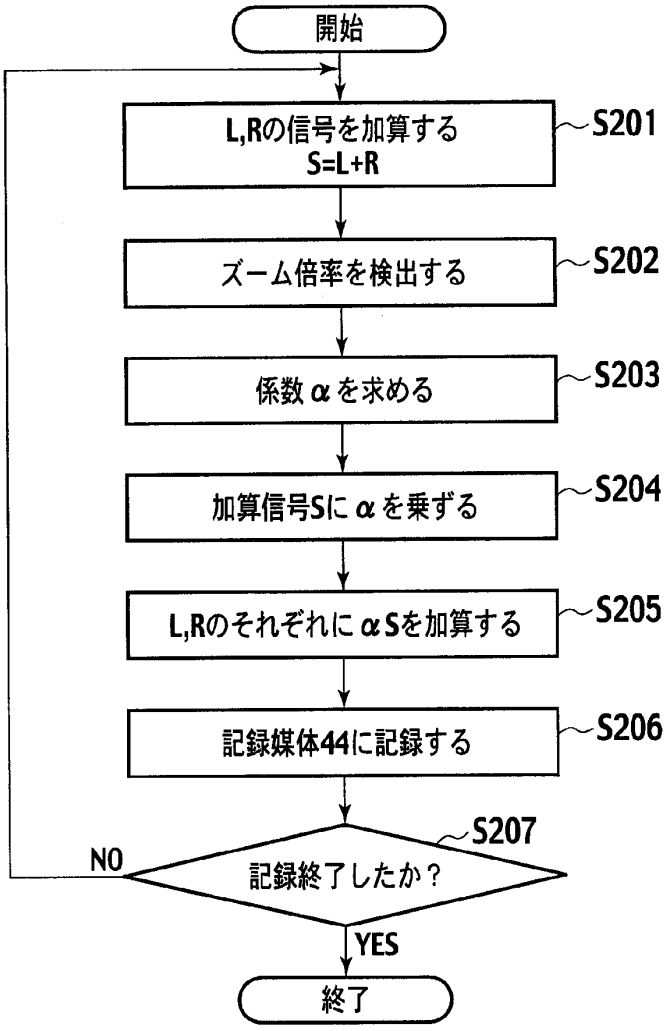
[図28]


59
53
モニタ
52
54
51
アンプ
37a
102
17
表示部
クロストークキャンセラ
d
Sw2
37b
48
操作部
47
27
c
制御部
Sw3
g
h
ビデオ復号化部
26
オーディオ復号化部
36
フラグ情報取得部
回路各部へ
16
15
分離化部
44
14
記録・再生部
記録媒体
13
多重化部
12
ビデオ符号化部
22
オーディオ符号化部
42
フラグ情報生成部
a
Sw1
b
41
接続検出部
11
カメラ部
21
内蔵ステレオマイク
33
音声ズーム処理部
32
3
31b
31a
302
302
30
300

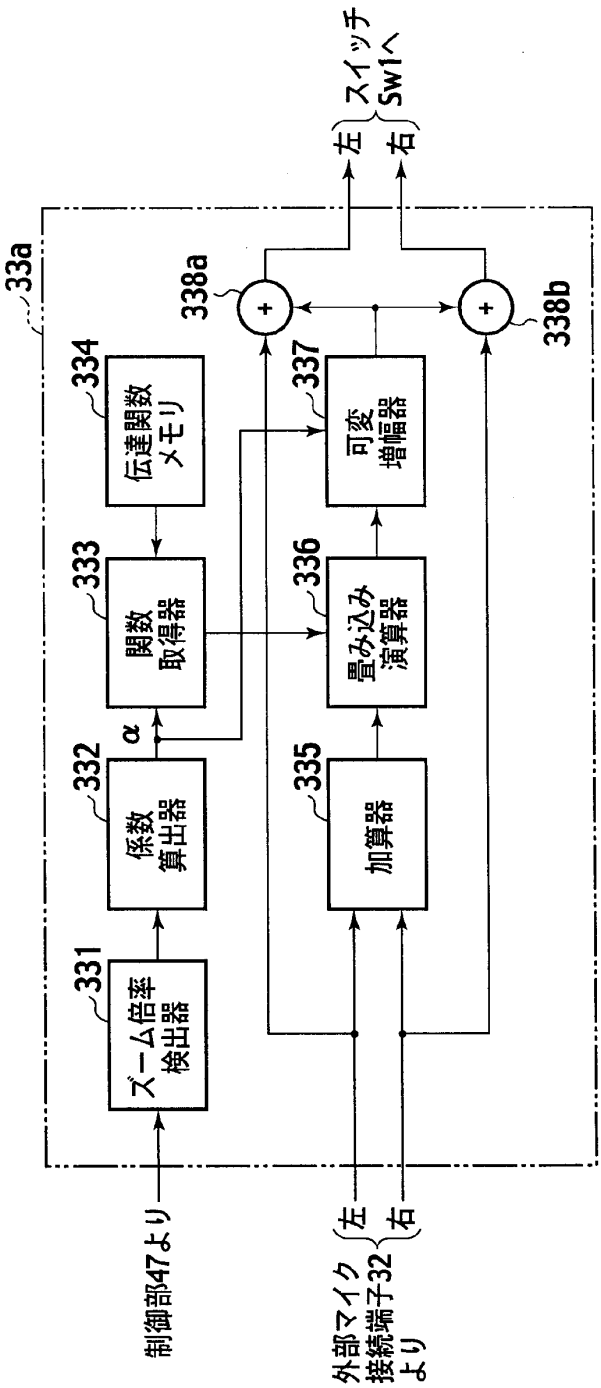
[図29]


33
331
332
制御部47から
ズーム倍率
検出器
係数
算出器
338a
α
335
337
外部マイク
接続端子32
から
左
右
加算器
可変
増幅器
左
右
スイッチ
Sw1へ
338b

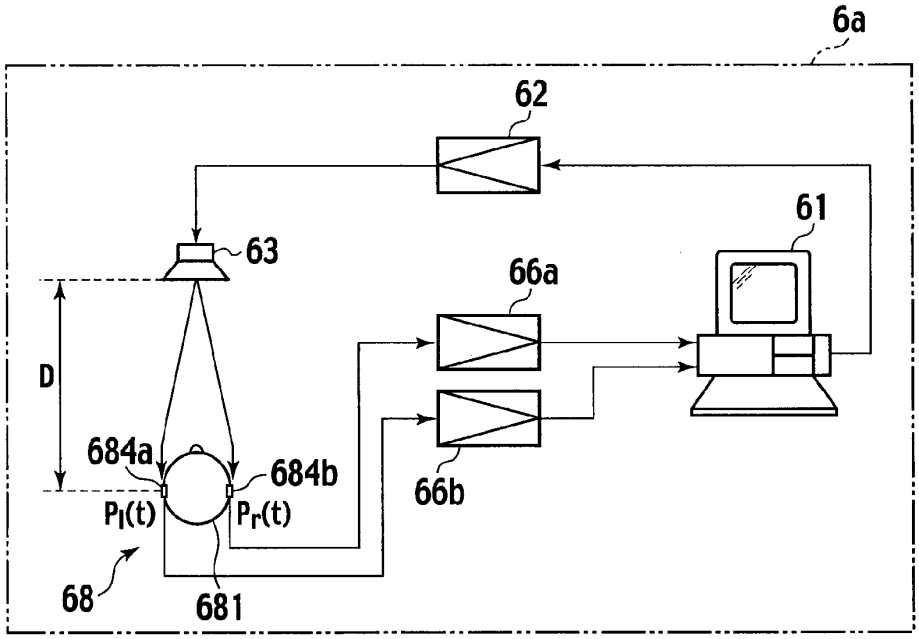
[図30]


開始
L,Rの信号を加算する
S=L+R
S201
ズーム倍率を検出する
S202
係数αを求める
S203
加算信号Sにαを乗ずる
S204
L,Rのそれぞれにα Sを加算する
S205
記録媒体44に記録する
S206
S207
記録終了したか？
NO
YES
終了

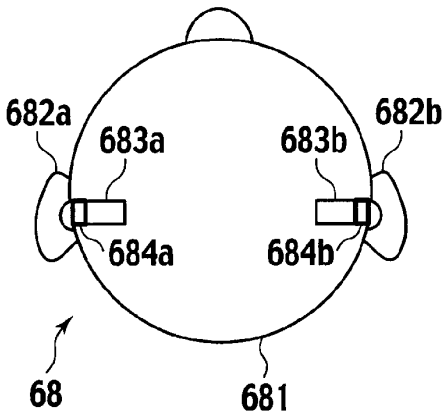
[図31]


33a
331
ズーム倍率検出器
332
係数算出器
α
333
関数取得器
334
伝達関数メモリ
335
加算器
336
畳み込み演算器
337
可変増幅器
338a
338b
+
+
左
右
スイッチSw1へ
制御部47より
外部マイク接続端子32より
左
右

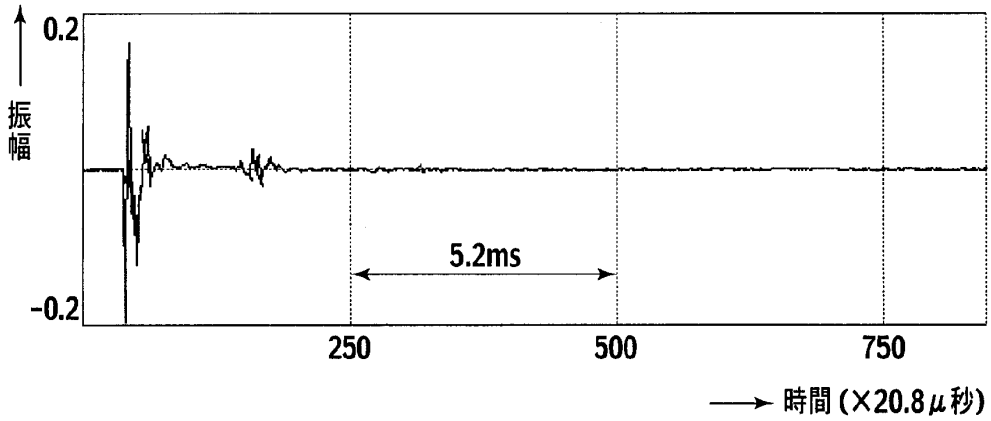
[図32]



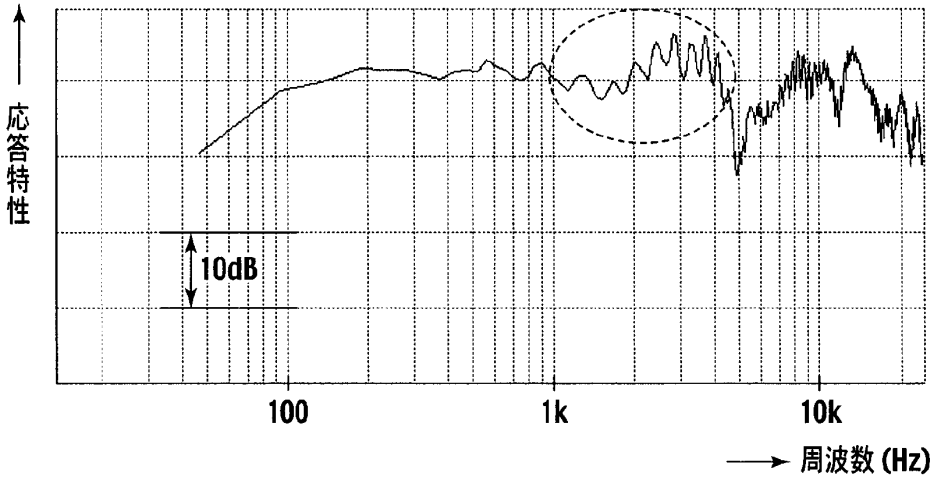
[図33]



[図34]



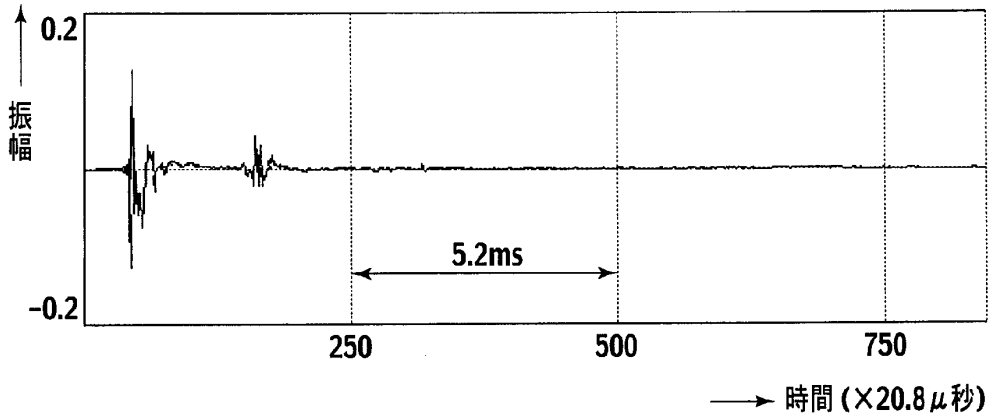
(A) インパルス応答波形



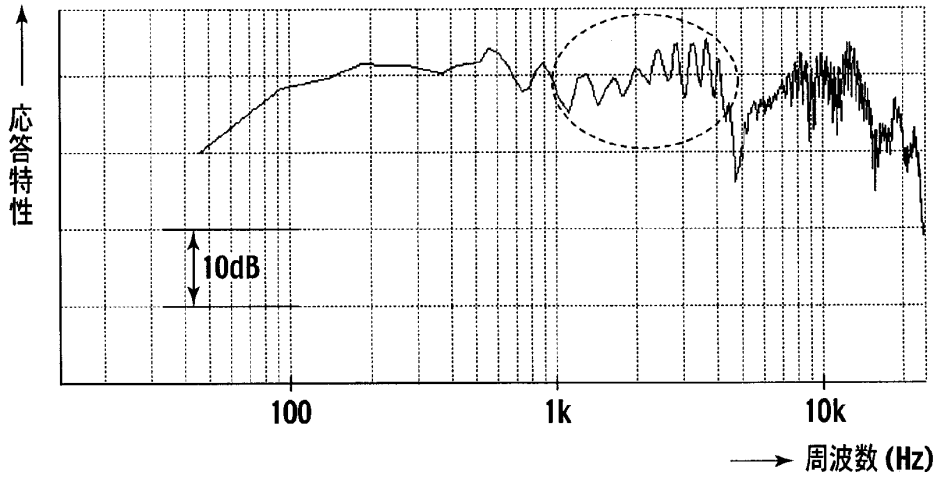
(B) 周波数応答特性

頭部伝達関数hl (t) D=0.5m

[図35]



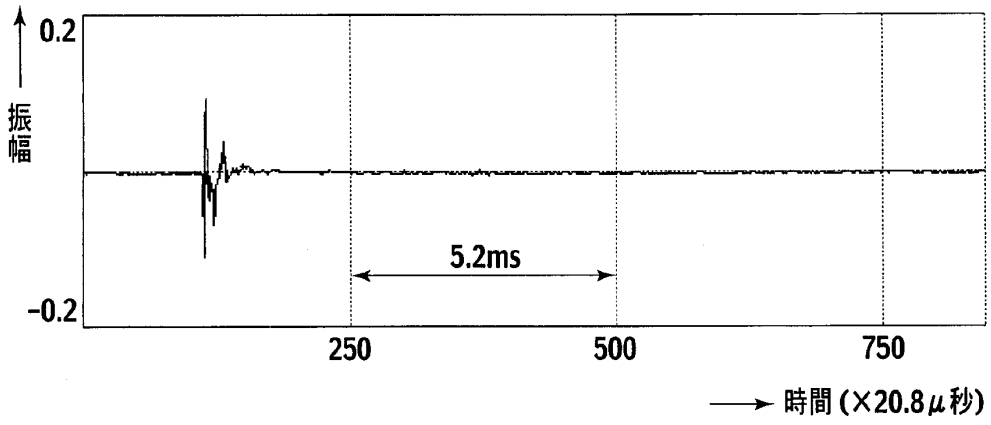
(A) インパルス応答波形



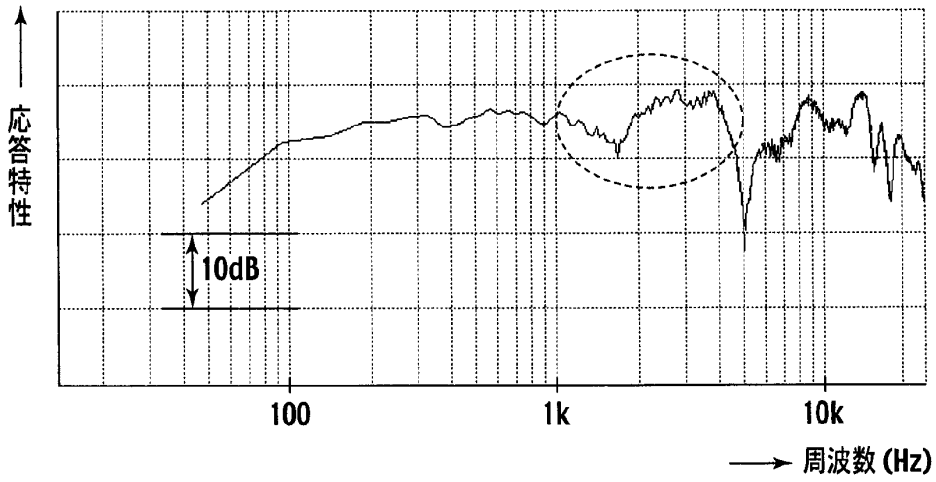
(B) 周波数応答特性

頭部伝達関数hr (t) D=0.5m

[図36]



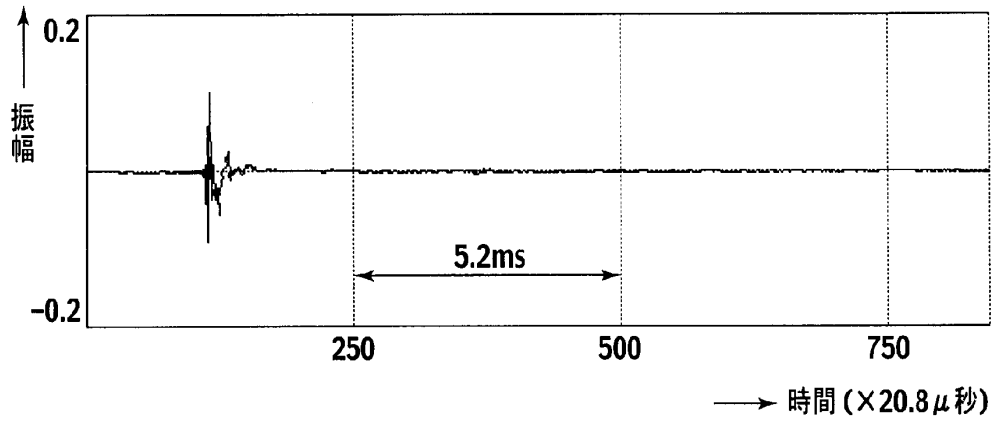
(A) インパルス応答波形



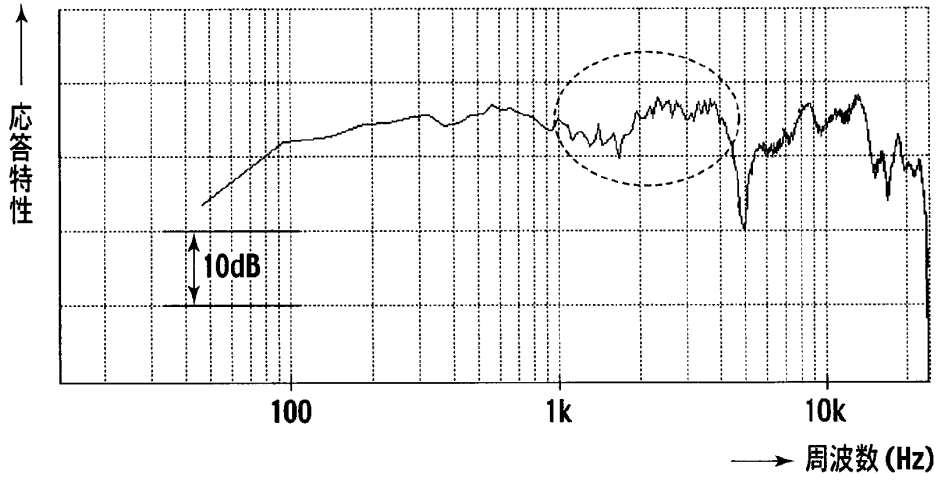
(B) 周波数応答特性

頭部伝達関数hl(t) D=1m

[図37]



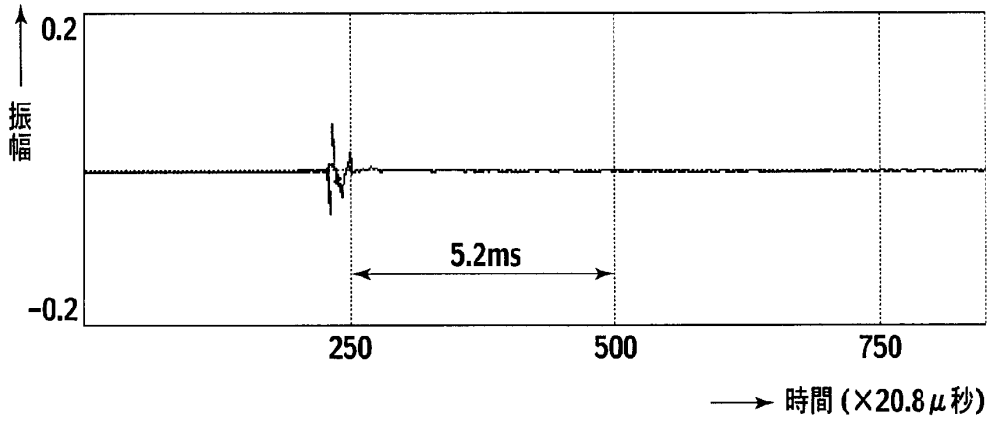
(A) インパルス応答波形



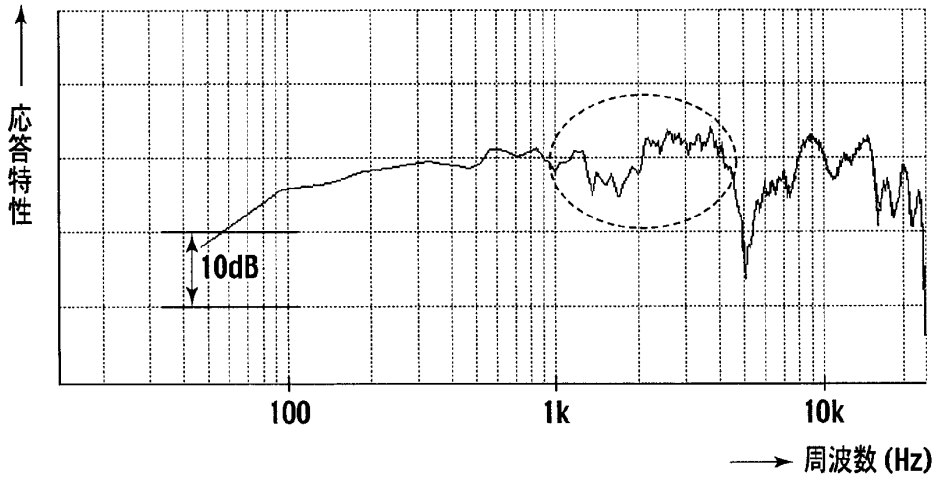
(B) 周波数応答特性

頭部伝達関数hr (t) D=1m

[図38]



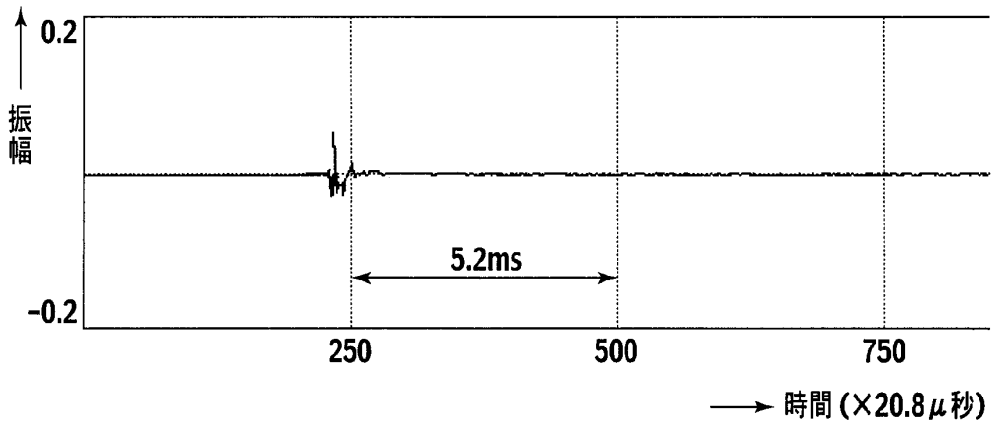
(A) インパルス応答波形



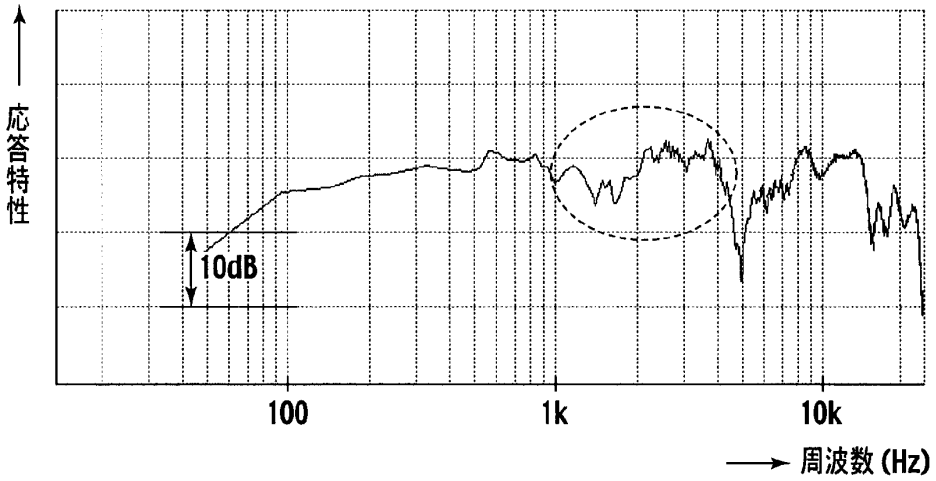
(B) 周波数応答特性

頭部伝達関数hl (t) D=2m

[図39]



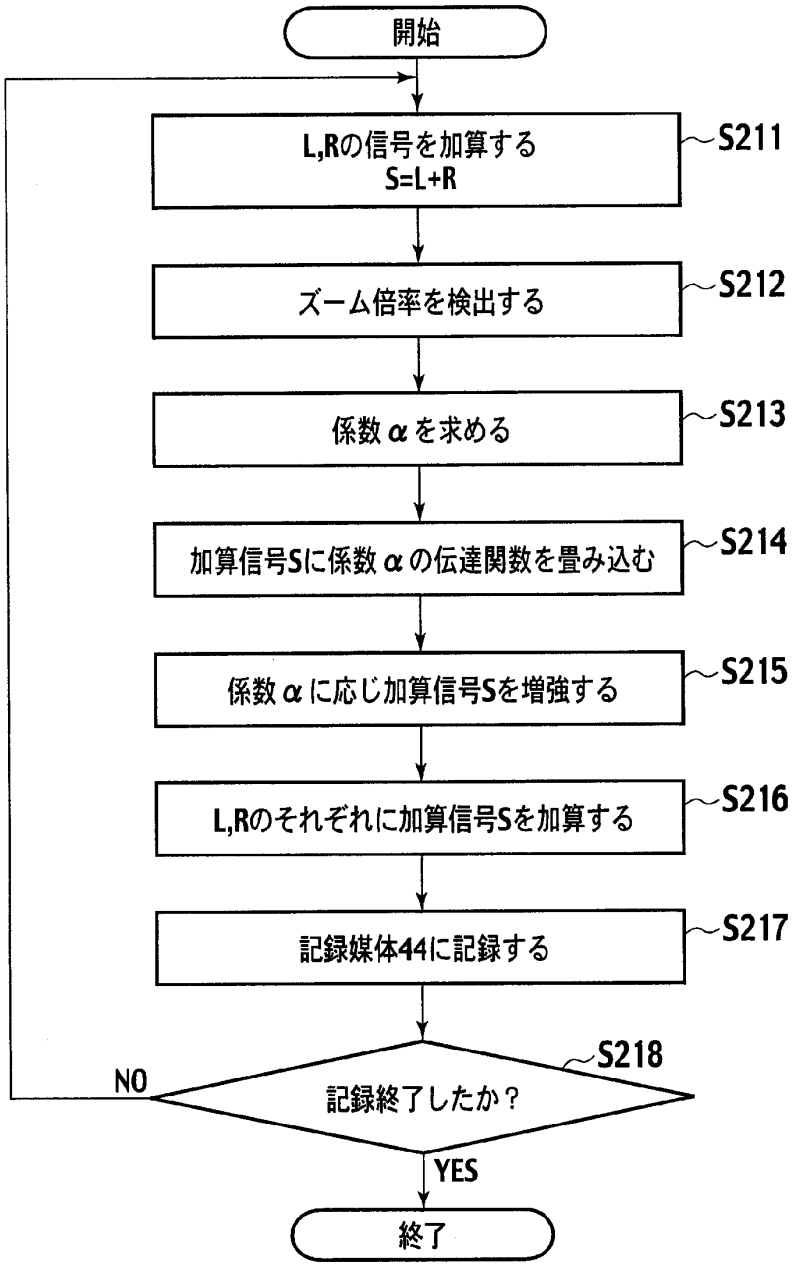
(A) インパルス応答波形



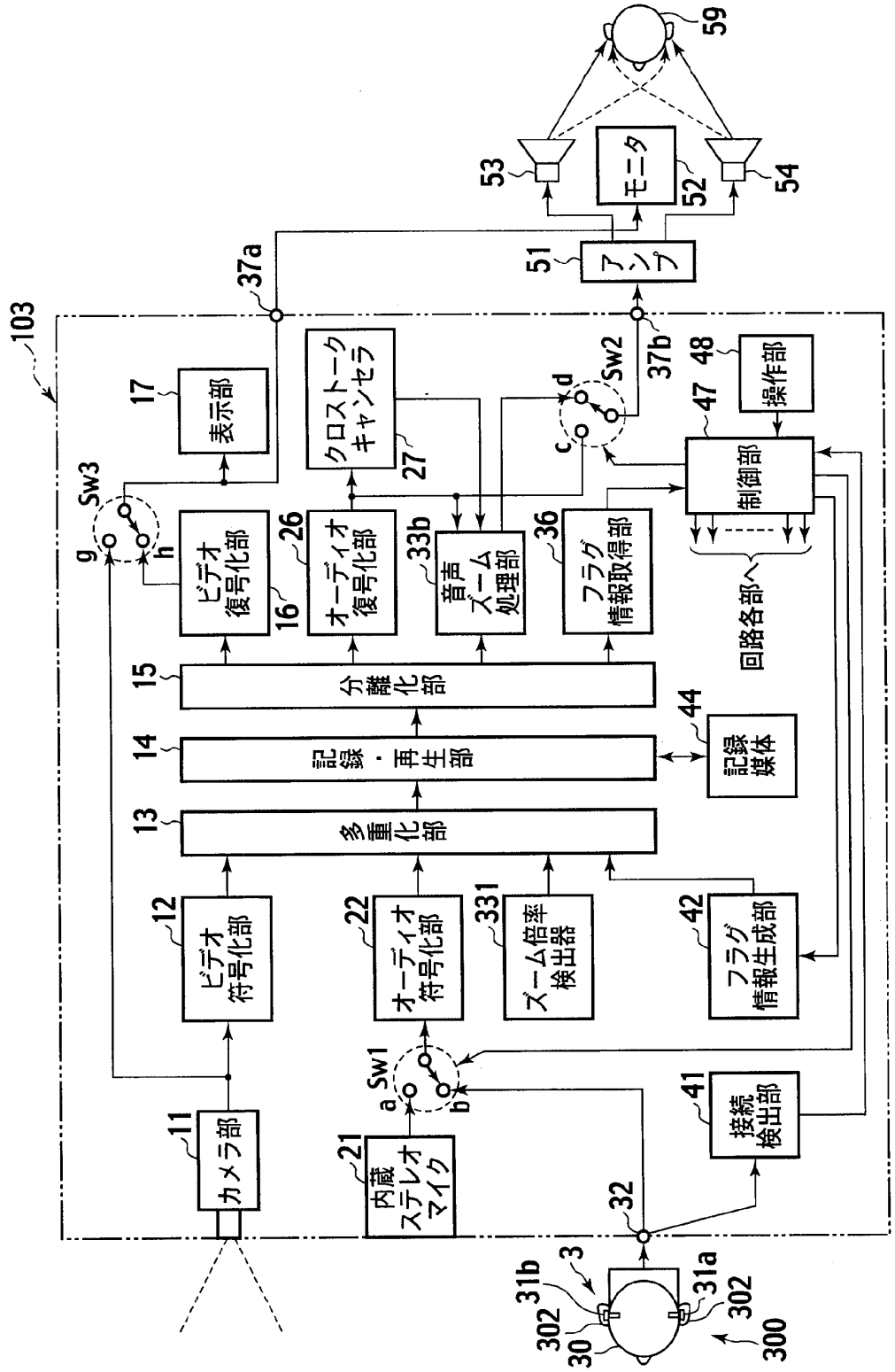
(B) 周波数応答特性

頭部伝達関数hr (t) D=2m

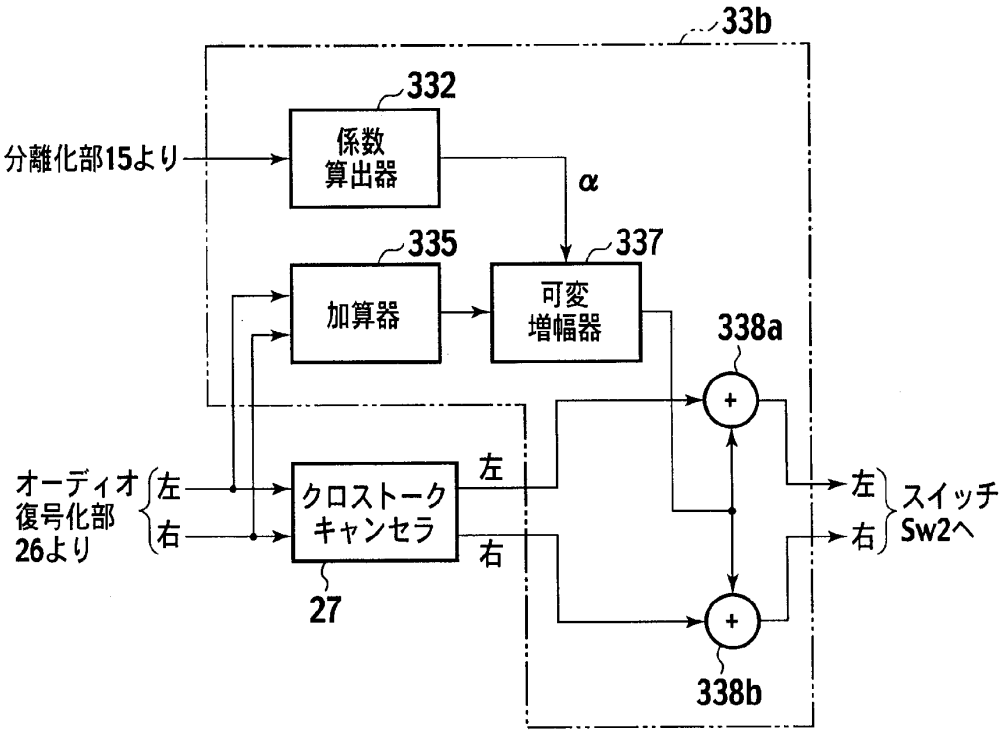
[図40]


開始
L,Rの信号を加算する
S=L+R
S211
ズーム倍率を検出する
S212
係数αを求める
S213
加算信号Sに係数αの伝達関数を畳み込む
S214
係数αに応じ加算信号Sを増強する
S215
L,Rのそれぞれに加算信号Sを加算する
S216
記録媒体44に記録する
S217
記録終了したか？
S218
NO
YES
終了

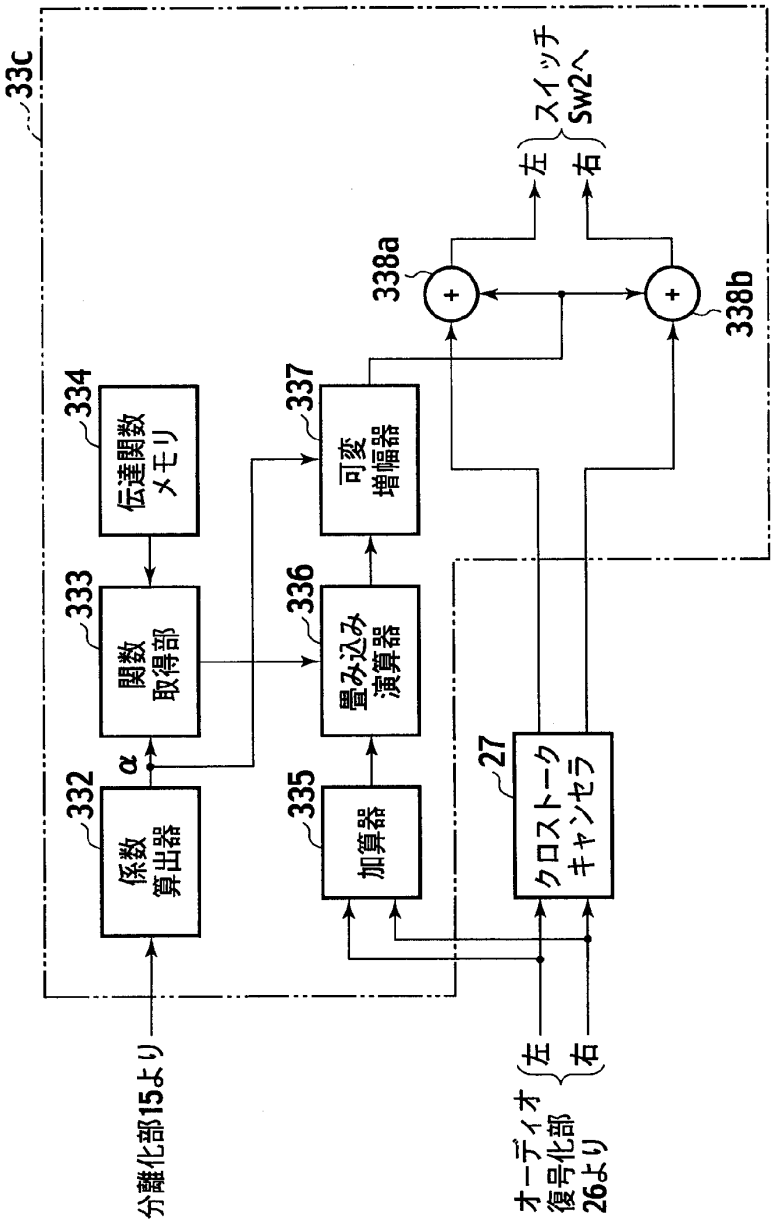
[図41]


59
53
モニタ
52
54
51
アンプ
37a
37b
103
17
表示部
クロストークキャンセラ
27
Sw2
d
c
48
操作部
47
制御部
Sw3
g
h
ビデオ復号化部
16
26
オーディオ復号化部
33b
音声ズーム処理部
36
フラグ情報取得部
回路各部へ
15
分離化部
44
記録媒体
14
記録・再生部
13
多重化部
12
ビデオ符号化部
22
オーディオ符号化部
331
ズーム倍率検出器
42
フラグ情報生成部
a
Sw1
b
41
接続検出部
11
カメラ部
21
内蔵ステレオマイク
32
3
31b
31a
302
30
300

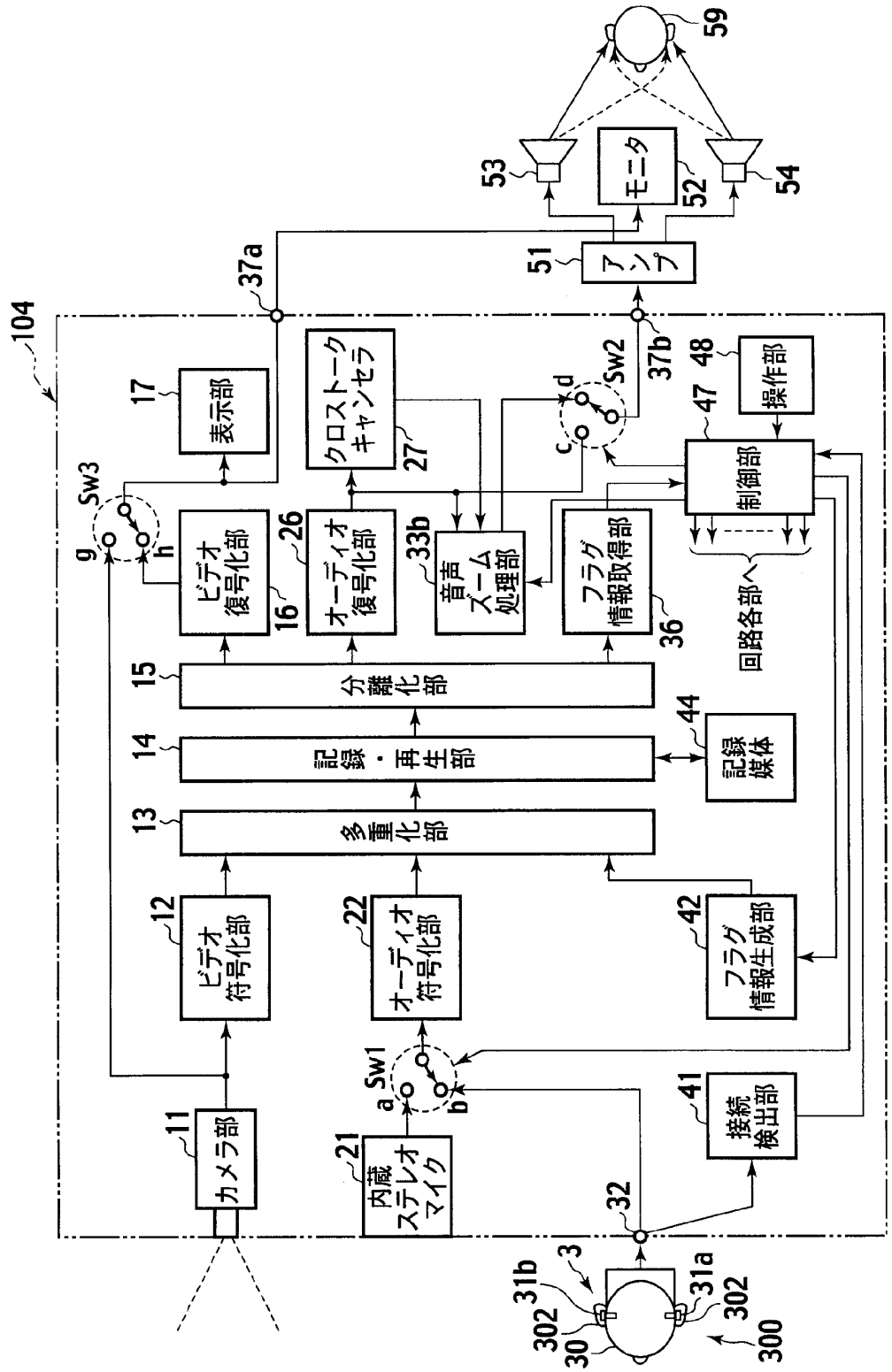
[図42]


33b
332
係数
算出器
分離化部15より
α
335
337
加算器
可変
増幅器
338a
オーディオ
復号化部
26より
左
右
クロストーク
キャンセラ
27
左
右
左
右
スイッチ
Sw2へ
338b

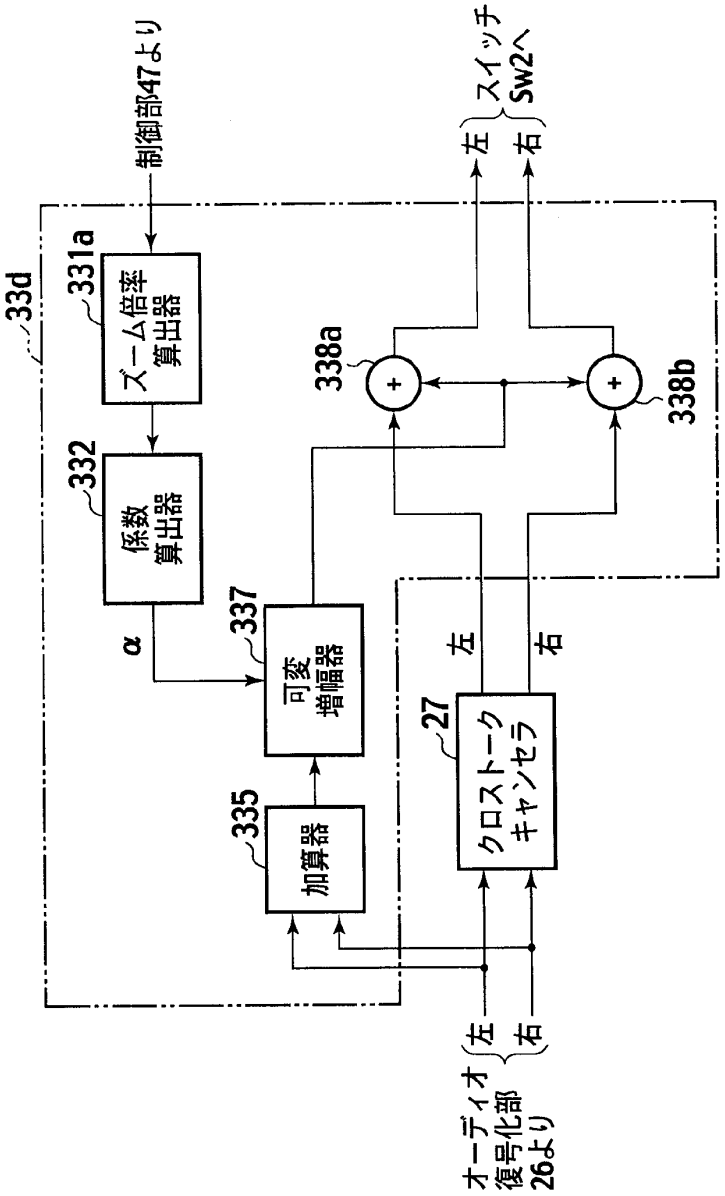
[図43]


33c
伝達関数メモリ
334
関数取得部
333
係数算出器
332
α
可変増幅器
337
畳み込み演算器
336
加算器
335
クロストークキャンセラ
27
338a
338b
+
+
左
右
スイッチSw2へ
分離化部15より
オーディオ復号化部26より
左
右

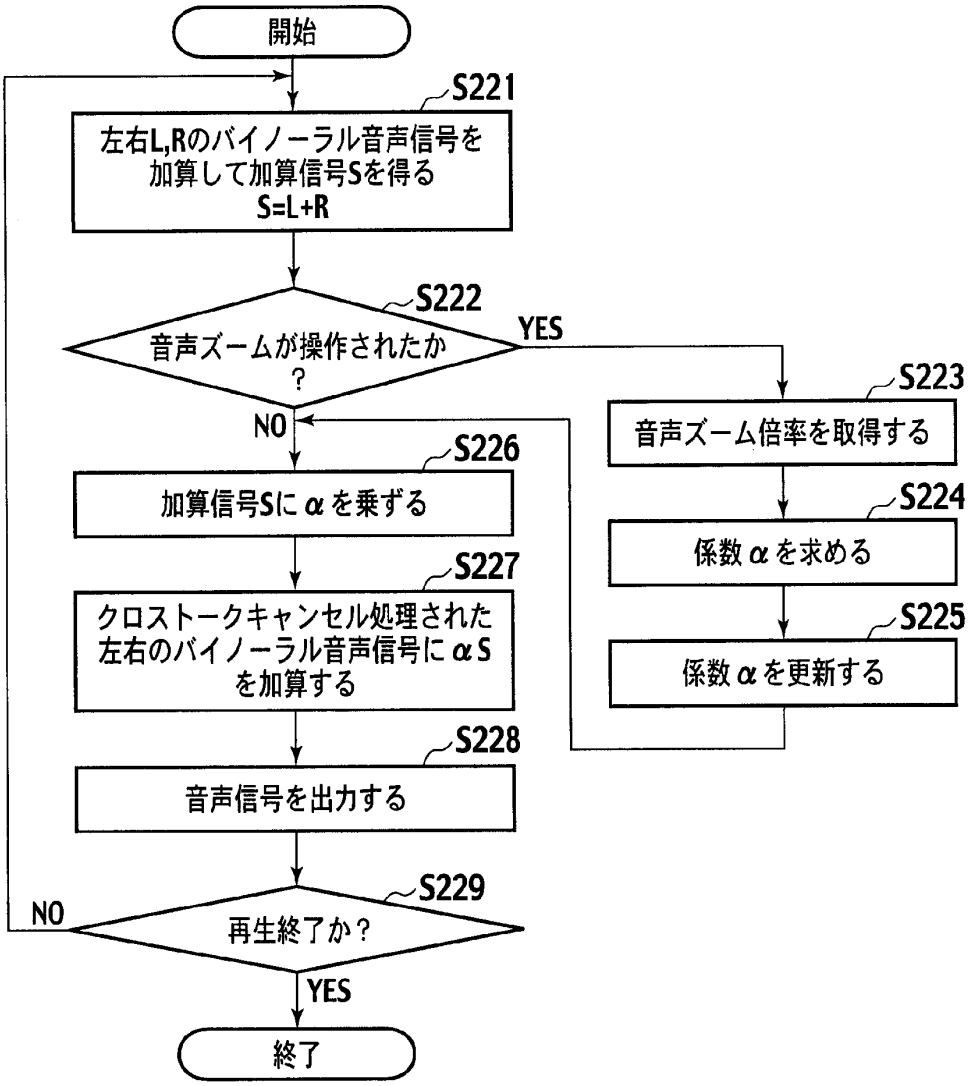
[図44]


59
53
モニタ
52
54
51
アンプ
37a
104
37b
48
操作部
17
表示部
クロストークキャンセラ
27
Sw2
d
c
47
制御部
Sw3
g
h
ビデオ復号化部
16
26
オーディオ復号化部
33b
音声ズーム処理部
フラグ情報取得部
36
回路各部へ
15
分離化部
44
記録媒体
14
記録・再生部
13
多重化部
12
ビデオ符号化部
22
オーディオ符号化部
42
フラグ情報生成部
a
Sw1
b
41
接続検出部
11
カメラ部
21
内蔵ステレオマイク
32
3
31b
31a
302
302
30
300

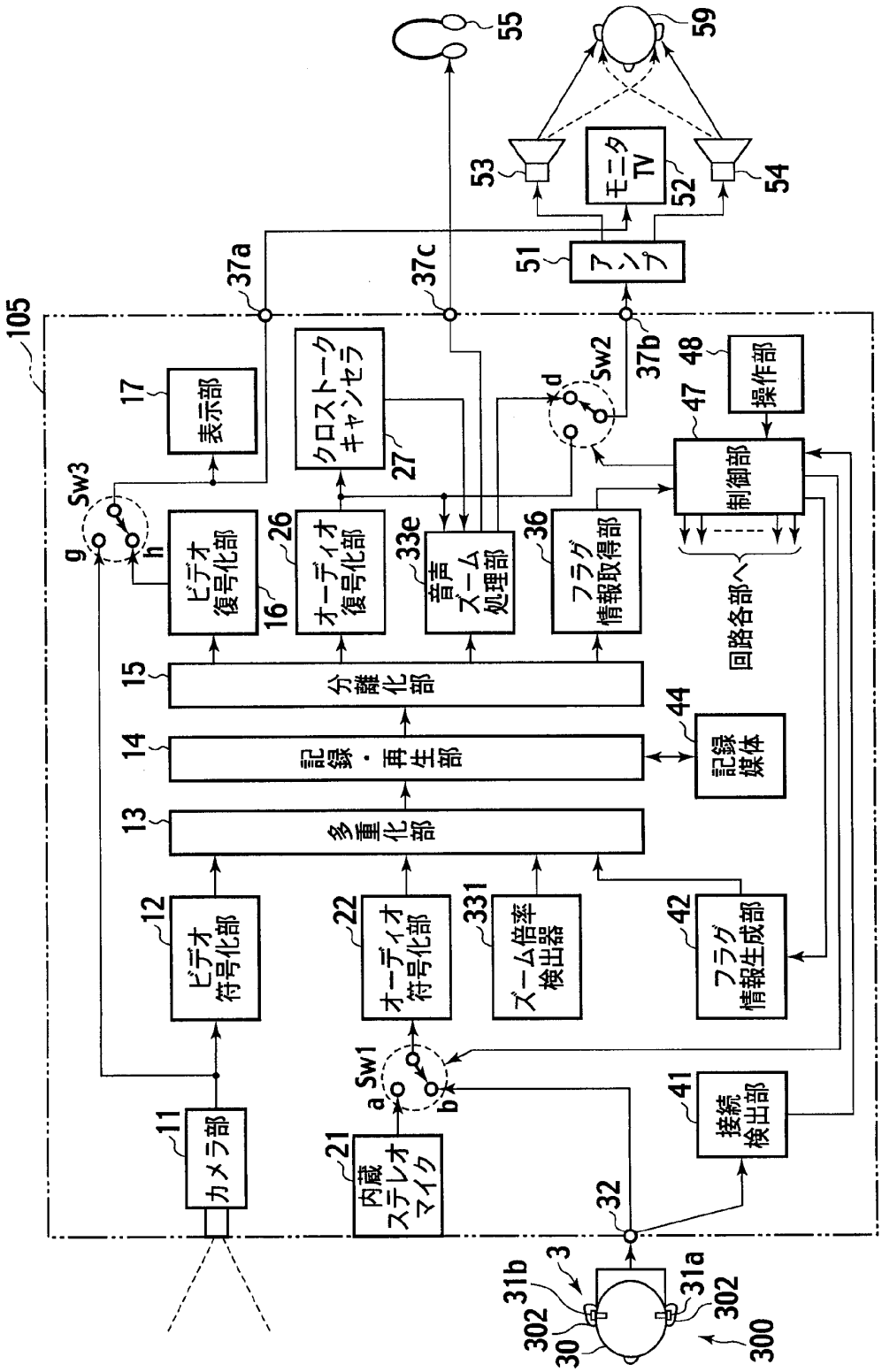
[図45]


制御部47より
33d
331a
ズーム倍率算出器
332
係数算出器
α
337
可変増幅器
335
加算器
338a
338b
27
クロストークキャンセラ
左
右
スイッチSw2へ
オーディオ復号化部26より

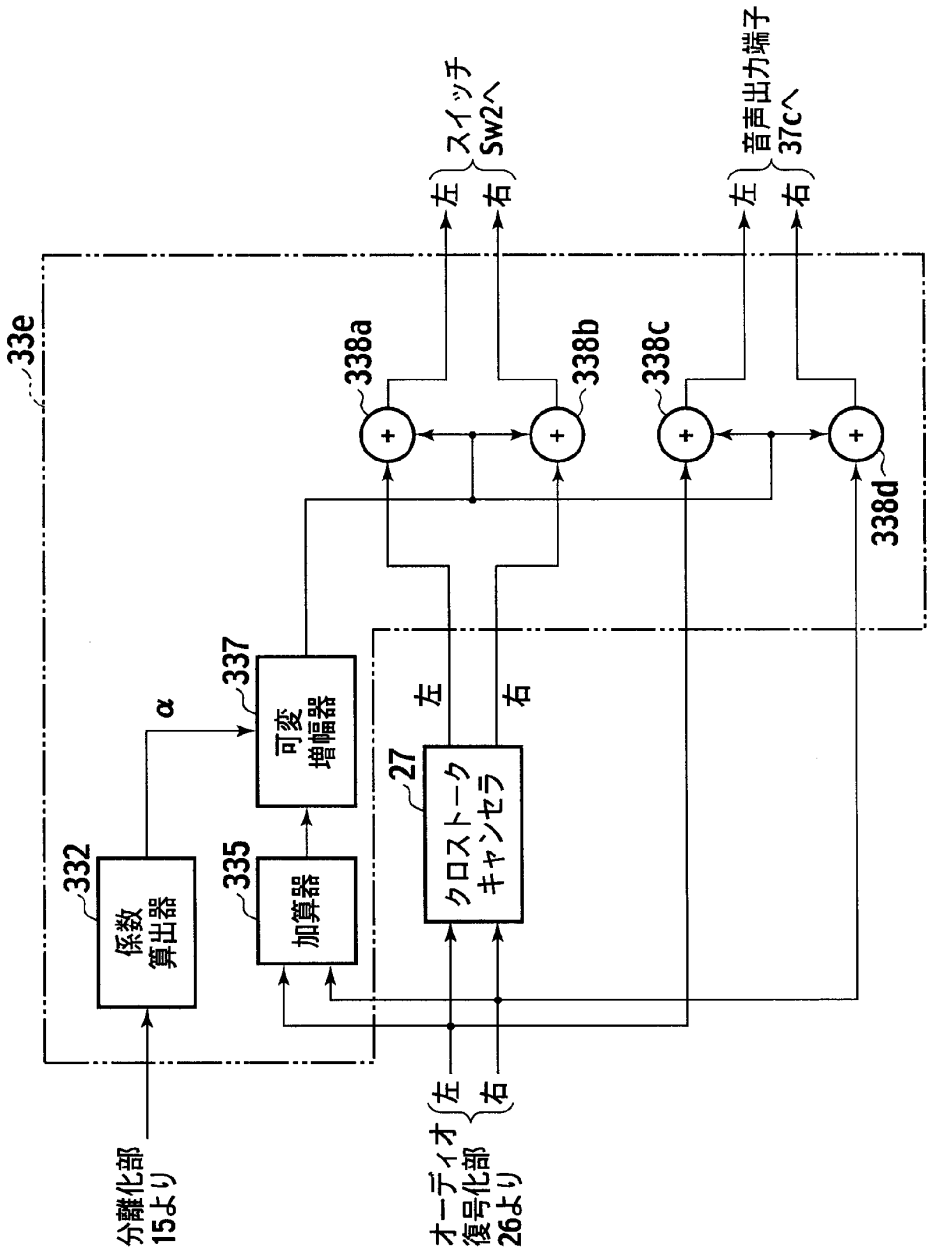
[図46]


開始
S221
左右L,Rのバイノーラル音声信号を
加算して加算信号Sを得る
S=L+R
S222
音声ズームが操作されたか
？
YES
NO
S223
音声ズーム倍率を取得する
S224
係数αを求める
S225
係数αを更新する
S226
加算信号Sにαを乗ずる
S227
クロストークキャンセル処理された
左右のバイノーラル音声信号にαS
を加算する
S228
音声信号を出力する
S229
再生終了か？
NO
YES
終了

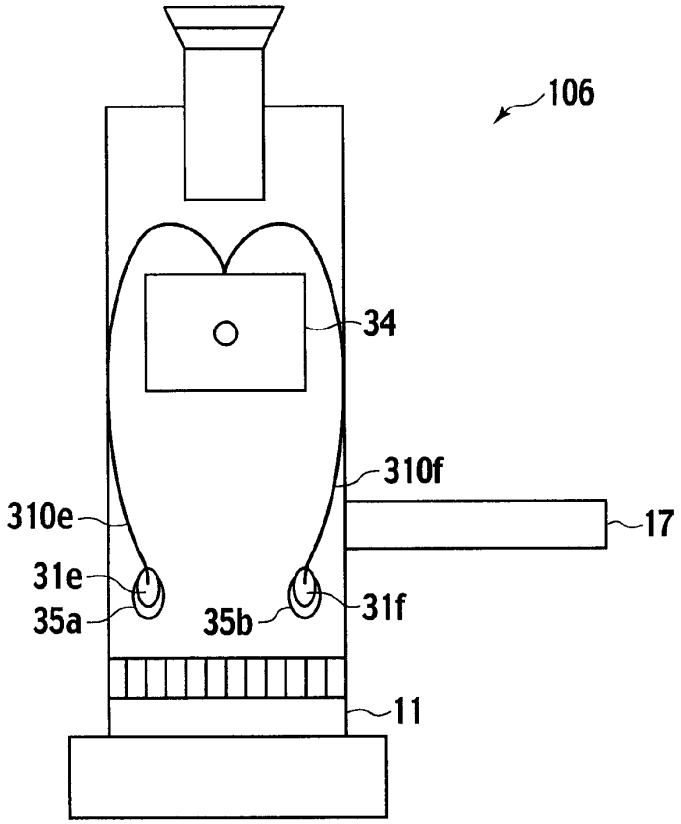
[図47]


105
37a
37c
37b
55
59
53
モニタTV
52
54
51
アンプ
17
表示部
クロストークキャンセラ
27
Sw2
d
48
操作部
47
制御部
Sw3
g
h
ビデオ復号化部
26
オーディオ復号化部
33e
音声ズーム処理部
36
フラグ情報取得部
回路各部へ
15
分離化部
44
記録媒体
14
記録・再生部
13
多重化部
12
ビデオ符号化部
22
オーディオ符号化部
331
ズーム倍率検出器
42
フラグ情報生成部
a
Sw1
b
41
接続検出部
11
カメラ部
21
内蔵ステレオマイク
32
31b
3
31a
302
302
30
300

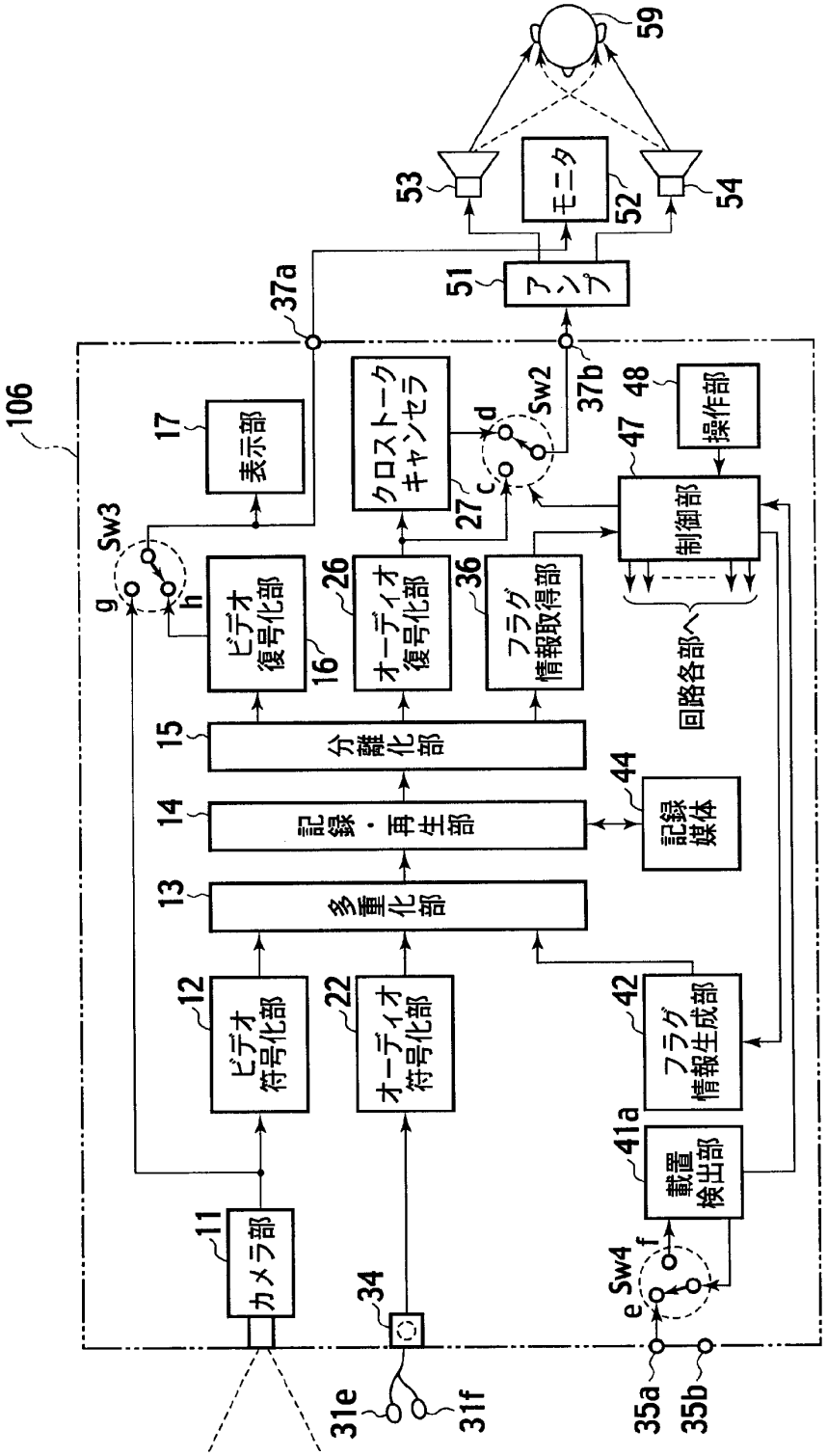
[図48]


33e
338a
338b
338c
338d
左
右
スイッチ
Sw2へ
音声出力端子
37cへ
337
可変
増幅器
α
27
クロストーク
キャンセラ
332
係数
算出器
335
加算器
分離化部
15より
オーディオ
復号化部
26より

[図49]

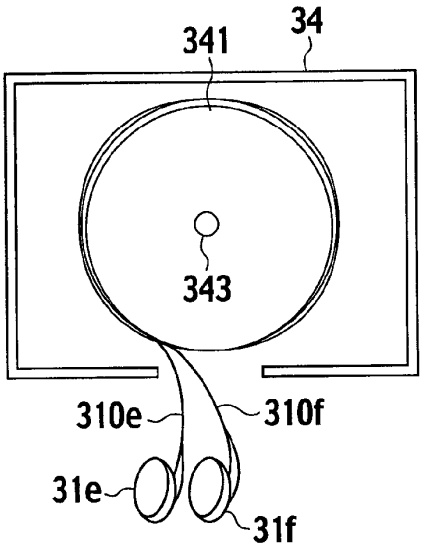


[図50]

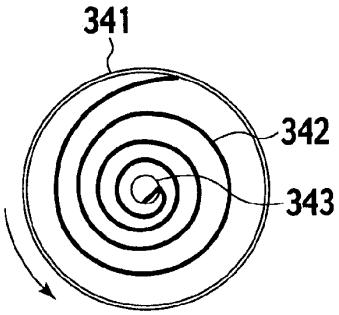

106
カメラ部
11
ビデオ符号化部
12
オーディオ符号化部
22
多重化部
13
記録・再生部
14
分離化部
15
ビデオ復号化部
16
オーディオ復号化部
26
フラグ情報取得部
36
表示部
17
クロストークキャンセラ
27
制御部
47
操作部
48
回路各部へ
フラグ情報生成部
42
載置検出部
41a
記録媒体
44
アンプ
51
モニタ
52
53
54
59
37a
37b
Sw1
Sw2
Sw3
Sw4
a
b
c
d
e
f
g
h
31e
31f
34
35a
35b

[図51]

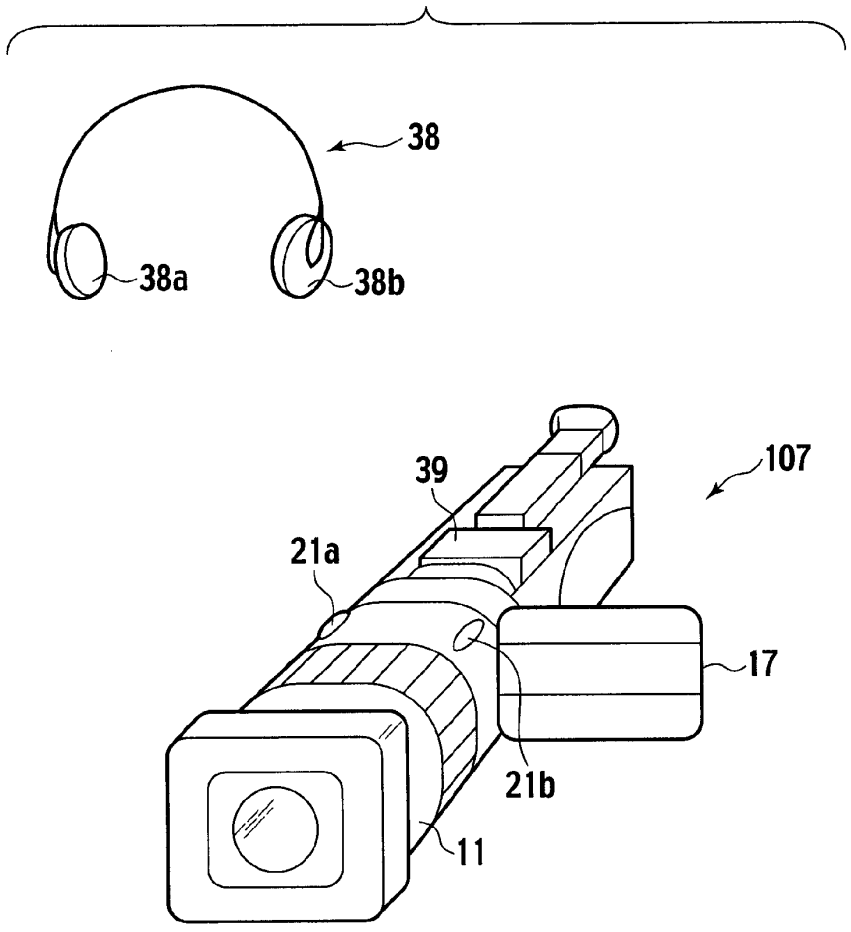
(A)



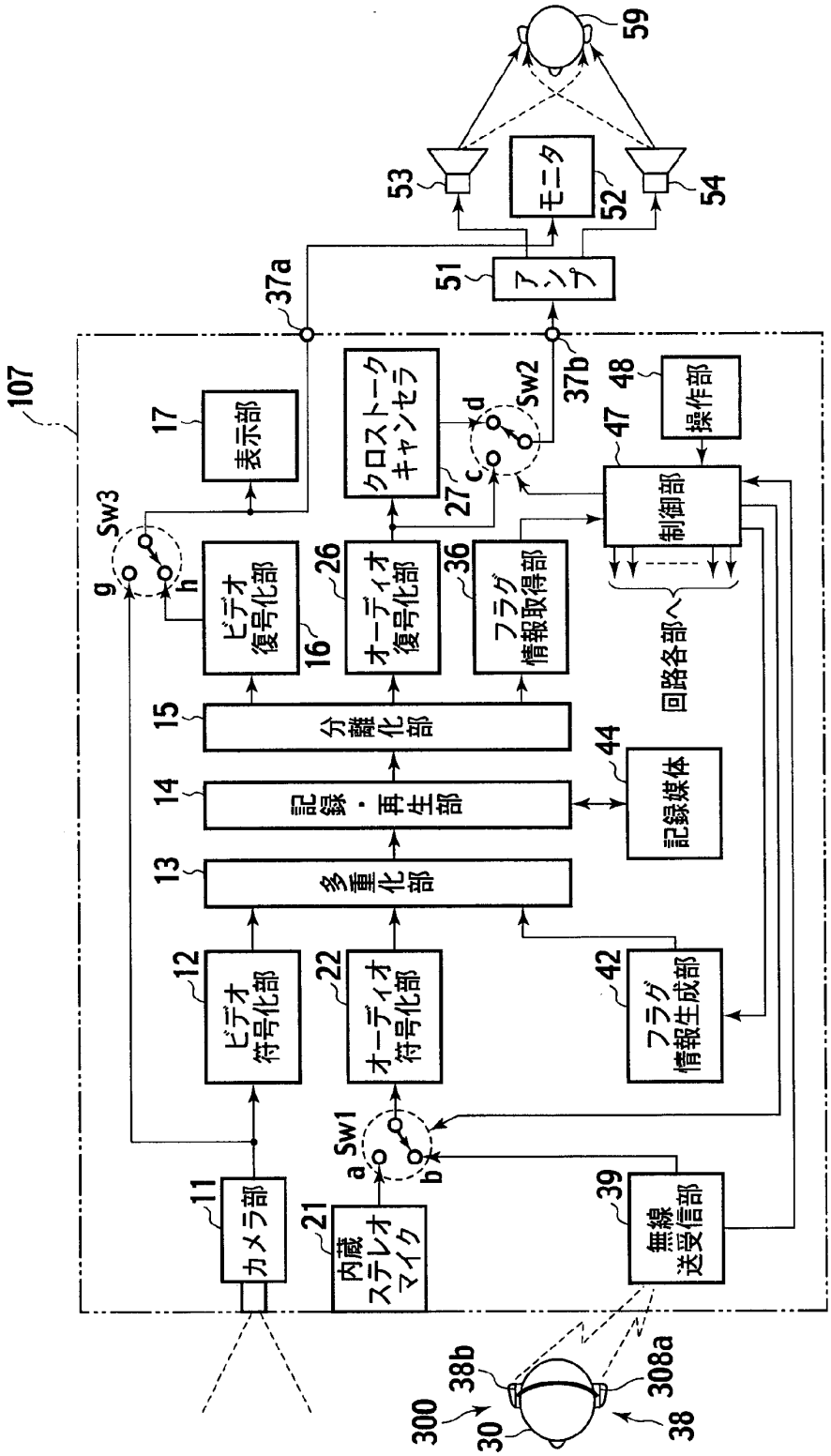
(B)



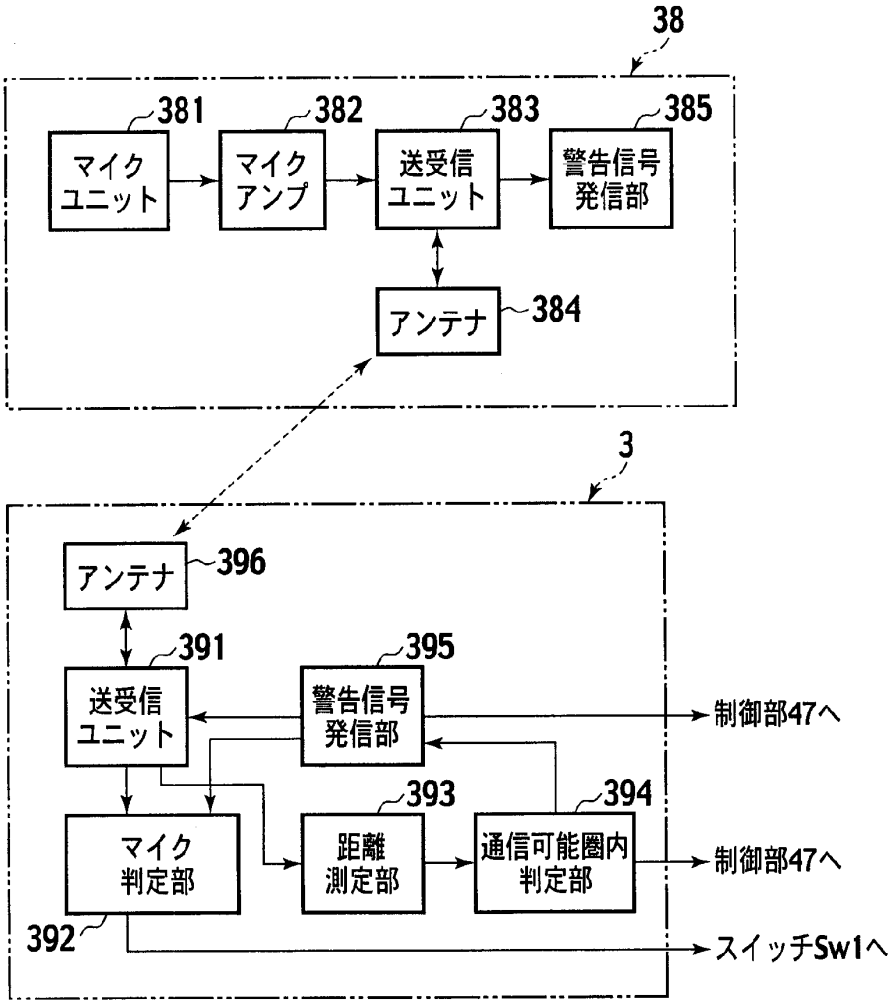
[図52]



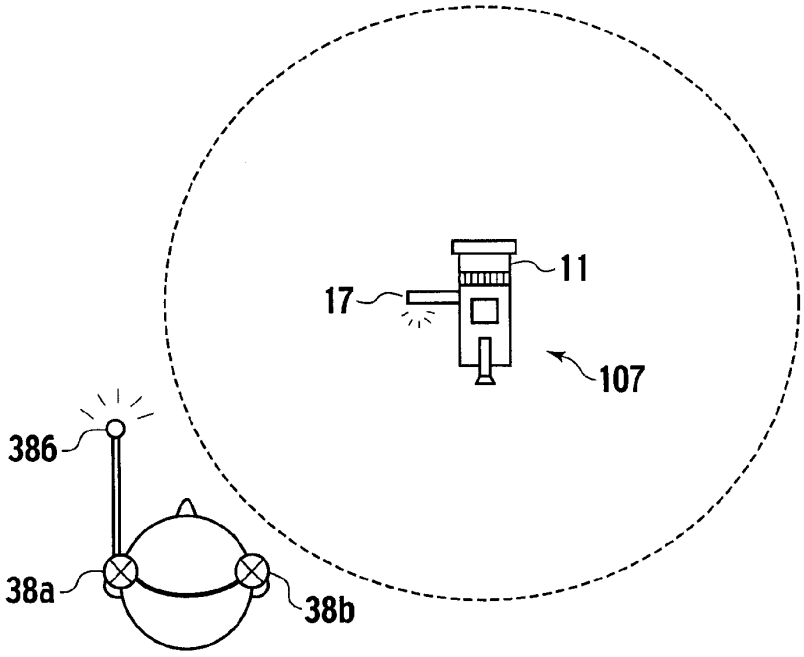
[図53]


59
53
モニタ
52
54
51
アンプ
37a
107
17
表示部
クロストークキャンセラ
Sw2
37b
48
操作部
47
制御部
27
Sw3
g
h
ビデオ復号化部
16
26
オーディオ復号化部
36
フラグ情報取得部
回路各部へ
15
分離化部
44
記録媒体
14
記録・再生部
13
多重化部
12
ビデオ符号化部
22
オーディオ符号化部
42
フラグ情報生成部
a
Sw1
b
11
カメラ部
21
内蔵ステレオマイク
39
無線送受信部
300
38b
30
308a
38

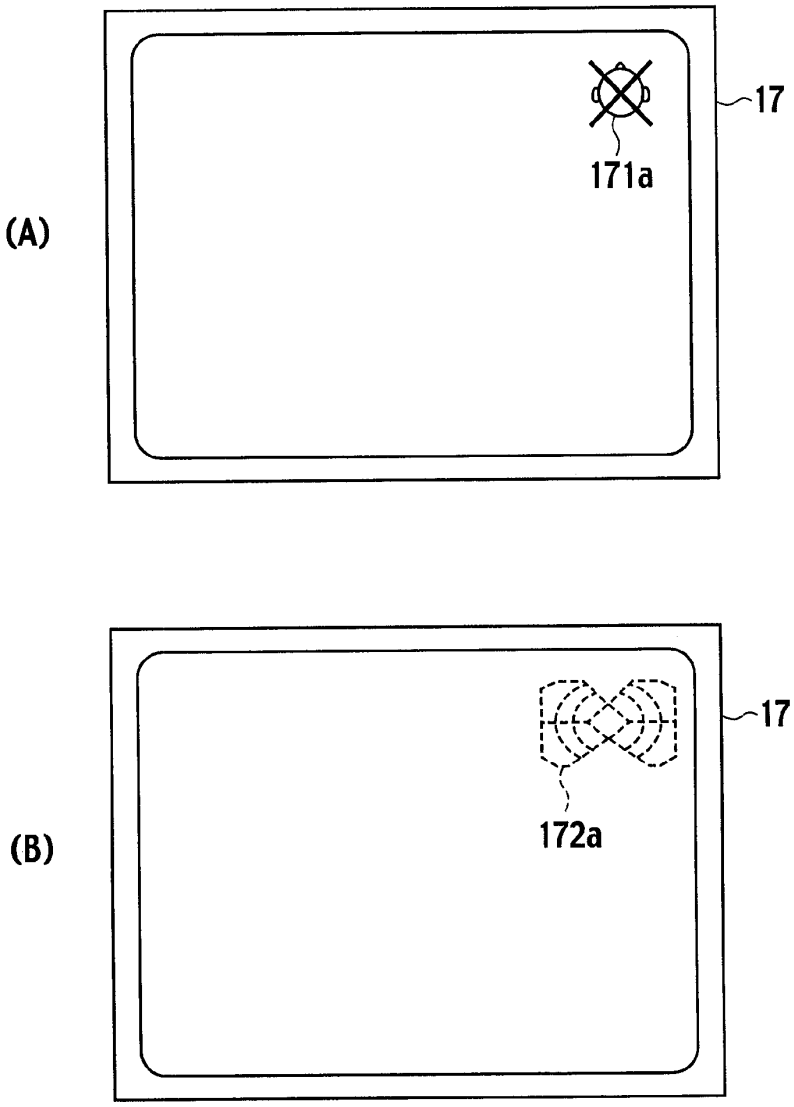
[図54]


38
381
マイク
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アンテナ
384
3
アンテナ
396
391
送受信
ユニット
395
警告信号
発信部
制御部47へ
393
マイク
判定部
距離
測定部
394
通信可能圏内
判定部
制御部47へ
392
スイッチSw1へ

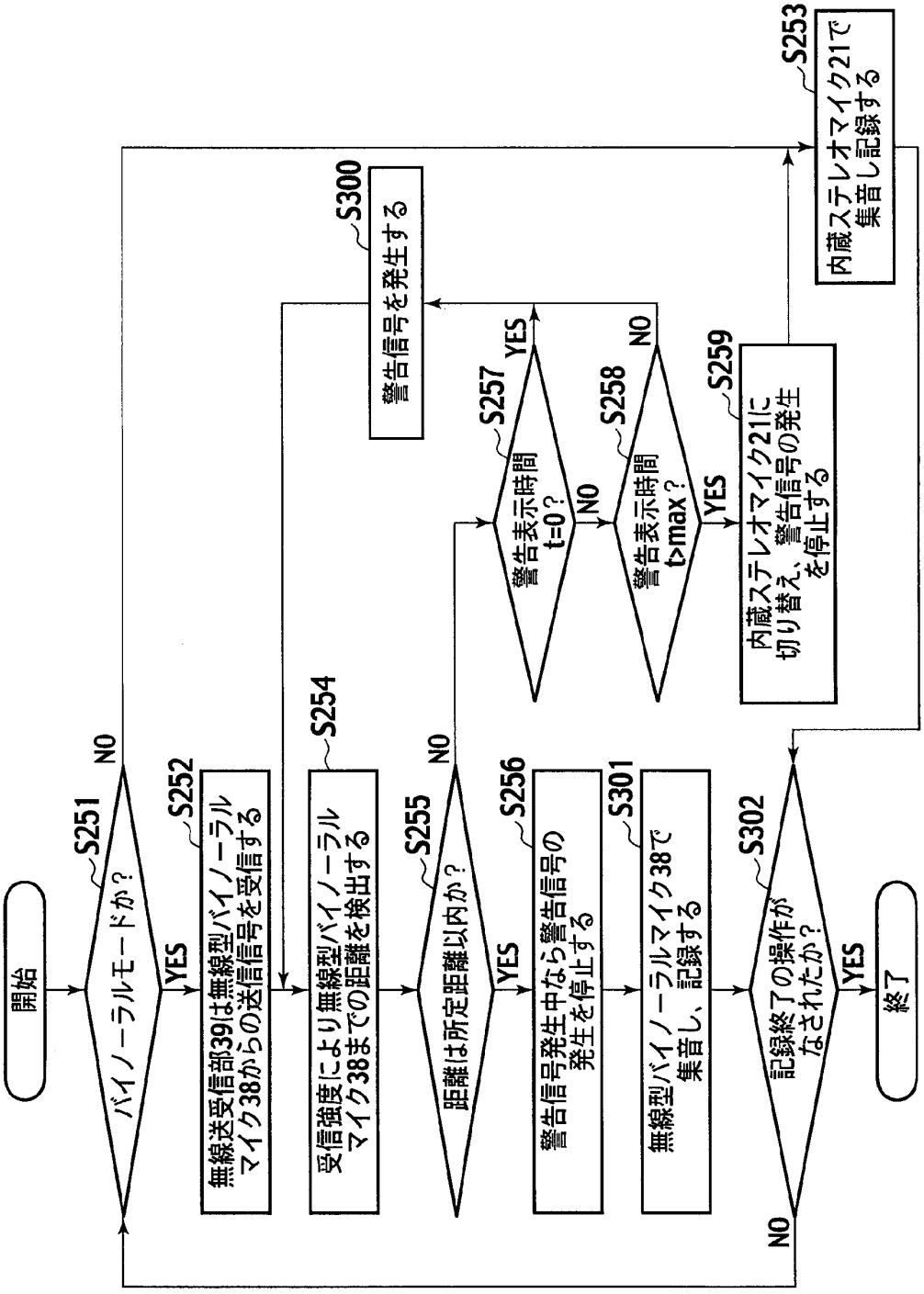
[図55]



[図56]



[図57]


開始
S251
バイノーラルモードか？
NO
YES
S252
無線送受信部39は無線型バイノーラルマイク38からの送信信号を受信する
S254
受信強度により無線型バイノーラルマイク38までの距離を検出する
S255
距離は所定距離以内か？
NO
YES
S256
警告信号発生中なら警告信号の発生を停止する
S301
無線型バイノーラルマイク38で集音し、記録する
S302
記録終了の操作がなされたか？
NO
YES
終了
S300
警告信号を発生する
S257
警告表示時間 t=0？
YES
NO
S258
警告表示時間 t>max？
NO
YES
S259
内蔵ステレオマイク21に切り替え、警告信号の発生を停止する
S253
内蔵ステレオマイク21で集音し記録する

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP2005/021247

A. CLASSIFICATION OF SUBJECT MATTER
H04S5/02(2006.01), ***H04S1/00***(2006.01), ***H04S7/00***(2006.01)

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)
H04S5/02(2006.01), ***H04S1/00***(2006.01), ***H04S7/00***(2006.01)

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
Jitsuyo Shinan Koho 1922-1996 Jitsuyo Shinan Toroku Koho 1996-2006
Kokai Jitsuyo Shinan Koho 1971-2006 Toroku Jitsuyo Shinan Koho 1994-2006

Electronic data base consulted during the international search (name of data base and, where practicable, 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.
Y	JP 5-244683 A (Sony Corp.), 21 September, 1993 (21.09.93), Par. Nos. [0008] to [0013], [0028] to [0030] (Family: none)	1-8
Y	JP 11-296998 A (Pioneer Electronic Corp.), 29 October, 1999 (29.10.99), Par. Nos. [0272] to [0281] & EP 952578 A2 & EP 1104928 A2 & CN 1439960 A	1-8
Y	JP 2004-222131 A (Matsushita Electric Industrial Co., Ltd.), 05 August, 2004 (05.08.04), Par. Nos. [0012] to [0016] (Family: none)	1-6

☒ 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 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
03 February, 2006 (03.02.06)	21 February, 2006 (21.02.06)
Name and mailing address of the ISA/ Japanese Patent Office Facsimile No.	Authorized officer Telephone No.

Form PCT/ISA/210 (second sheet) (April 2005)

INTERNATIONAL SEARCH REPORT

international application No.
PCT/JP2005/021247

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y A	JP 7-312712 A (Sanyo Electric Co., Ltd.), 28 November, 1995 (28.11.95), Par. Nos. [0006] to [0008] (Family: none)	5 1-4,6-8
Y A	JP 7-336793 A (Matsushita Electric Industrial Co., Ltd.), 22 December, 1995 (22.12.95), Par. Nos. [0002] to [0009] (Family: none)	5 1-4,6
Y	JP 2002-165300 A (Sharp Corp.), 07 June, 2002 (07.06.02), Par. Nos. [0006] to [0012] (Family: none)	1-8
Y A	JP 10-70789 A (Central Research Laboratories Ltd.), 10 March, 1998 (10.03.98), Par. No. [0008] & US 5917916 A & GB 9610394 A & EP 808075 A	7-8 1-6

Form PCT/ISA/210 (continuation of second sheet) (April 2005)

国際調査報告	国際出願番号 PCT／JP2005／021247

A. 発明の属する分野の分類（国際特許分類（IPC））

Int.Cl. *H04S5/02 (2006.01), H04S1/00 (2006.01), H04S7/00 (2006.01)*

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. *H04S5/02 (2006.01), H04S1/00 (2006.01), H04S7/00 (2006.01)*

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	1971－2006年
日本国実用新案登録公報	1996－2006年
日本国登録実用新案公報	1994－2006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）

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	JP 5-244683 A（ソニー株式会社）1993.09.21, 【0008】-【0013】, 【0028】-【0030】（ファミリ無し）	1-8
Y	JP 11-296998 A（パイオニア株式会社）1999.10.29, 【0272】-【0281】 & EP 952578 A2 & EP 1104928 A2 & CN 1439960 A	1-8
Y	JP 2004-222131 A（松下電器産業株式会社）2004.08.05, 【0012】-【0016】（ファミリ無し）	1-6

☑ 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
03．02．2006	21．02．2006

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）	5Z	9072
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	井出 和水 電話番号 03－3581－1101 内線 3541		

様式PCT／ISA／210（第2ページ）（2005年4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 A	JP 7-312712 A（三洋電機株式会社）1995.11.28,　【0006】-【0008】 （ファミリ無し）	5 1-4,6-8
Y A	JP 7-336793 A（松下電器産業株式会社）1995.12.22,　【0002】-【0009】 （ファミリ無し）	5 1-4,6
Y	JP 2002-165300 A（シャープ株式会社）2002.06.07,　【0006】-【0012】 （ファミリ無し）	1-8
Y A	JP 10-70789 A（セントラル リサーチ ラボラトリーズ リミティド） 1998.03.10,　【0008】　& US 5917916 A & GB 9610394 A & EP 808075 A	7-8 1-6

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European Patent Office
Office européen des brevets

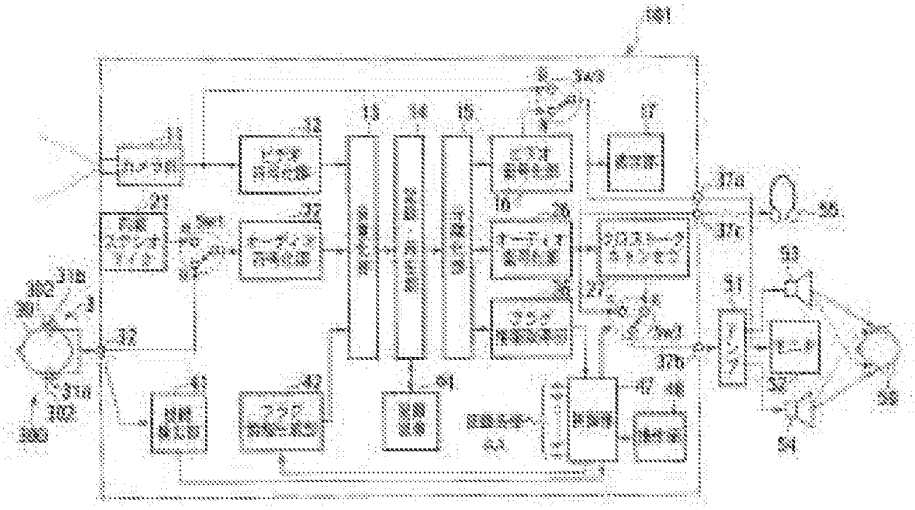
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Notice

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ABSTRACT JPWO2006054698A1

The video / audio recording / playback device (101) includes a built-in stereo microphone (21a, 21b) and an external microphone connection terminal (32). A binaural microphone (3) to be worn on the ear of the photographer (300) is connected to the external microphone connection terminal (32). When the ambient sound is collected by the binaural microphone (3), the audio signal to be recorded on the recording medium is switched from the audio signal from the built-in stereo microphones (21a, 21b) to the binaural audio signal from the binaural microphone (3). The photographer (300) wears a binaural microphone (3 (31a.31b)) in his ear and collects ambient sounds of the photographer (300) including sounds emitted from the subject. The camera unit (11) captures a subject. The binaural audio signal, the captured video signal, and the binaural flag signal are recorded on the recording medium.



11 CAMERA PART
21 INTERNALLY MOUNTED STEREO MICROPHONES
41 DISTINCTION DETECTING PART
12 VIDEO ENCODING PART
22 AUDIO ENCODING PART
42 FLAG INFORMATION GENERATING PART
13 MULTIPLEXING PART
14 RECORDING/REPRODUCING PART
44 RECORDING MEDIUM
15 SEPARATING PART
16 VIDEO DECODING PART
26 AUDIO DECODING PART
36 FLAG INFORMATION ACQUIRING PART
AA TO VARIOUS CIRCUIT COMPONENTS
17 DISPLAY PART
27 CROSSTALK CANCELLER
47 CONTROL PART
48 OPERATING PART
51 AMPLIFIER
52 MONITOR

WO2006054698A1 VIDEO/AUDIO RECORDING APPARATUS AND METHOD, AND VIDEO/AUDIO REPRODUCING APPARATUS AND METHOD

Data originating from sources other than the EPO may not be accurate, complete, or up to date.

The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

Specification

Video / audio recording apparatus and method, audio / video reproduction apparatus and method Technical field

[0001] TECHNICAL FIELD The present invention relates to a video and audio recording apparatus and method for recording a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer including a sound emitted from a subject, and an image and sound recording apparatus A video and audio reproducing apparatus and method for reproducing a signal and an audio signal, and more particularly to a video and audio recording apparatus and method, and a video and audio reproducing apparatus and method capable of reproducing realistic sound with a captured image .

BACKGROUND ART

[0002] Video / audio recording / reproducing apparatuses (1, so-called video cameras) are widely used to record the image signal of the subject and the sound signal collected by the photographer's surroundings including the sound emitted from the subject. This type of video / audio recording / reproducing apparatus has a stereo mic for recording sounds in a stereo manner. In recent years, the video / audio recording / reproducing apparatus has been further miniaturized, and there is a problem that it is difficult to record a realistic sound in a stereo microphone mounted on a compact video / audio recording / reproducing apparatus. Therefore, there is a demand for a video / audio recording / reproducing apparatus capable of recording a more realistic sound.

[0003] In International Publication No. 96Z10884 pamphlet records the binaural collected sound field of the sound field around the photographer together with the image signal of the object picked up by disposing the ear structure on both sides of the main body of the video and audio recording and reproducing apparatus Audio / video recording / reproducing apparatus capable of recording / reproducing video / audio data.

[0004] In the video / audio recording / reproducing apparatus in which the binaural microphone disclosed in the above-mentioned document is attached, the lateral width of the main unit of the apparatus, that is, the distance between the left and right microphones is not equal to that of the human head, I can not record sensible sounds. In recent years, the main body of video / audio recording / reproducing apparatus which has been commercialized has been miniaturized in accordance with the advancement of high density recording technology, digital signal recording technology and video compression processing technology so that video / audio recording / reproducing Even if a microphone intended for binaural sound collection is mounted on the main body of the apparatus, the effect expected can not be obtained. Further, since the shape of the apparatus is largely different from the human head, it is considered difficult to obtain the effect disclosed in the above document.

Disclosure of invention

[0005] SUMMARY OF THE INVENTION The present invention has been made in view of the above problems, and it is an object of the present invention to provide a video and audio recording apparatus and method capable of reproducing realistic voices with images captured irrespective of the size and shape of the apparatus main body, It is an object of the present invention to provide a video and audio reproducing apparatus and method

[0006] A video and audio recording apparatus and method capable of reproducing a more realistic sound in conjunction with zooming-up of a subject when zooming up the subject, and a video and audio reproducing apparatus and method It is aimed to provide.

[0007] Furthermore, even in the case where an arbitrary photographer shoots and watches an arbitrary viewer, as in the case where the viewer who views the reproduction signal and the photographer who captured the binaural

sound are different from each other, It is an object of the present invention to provide a video and audio reproducing apparatus and method capable of listening to a sensitive voice.

[0008] In order to solve the problem of the above-described conventional technique, the present invention provides an image recording apparatus for recording an image signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer including a sound emitted from the subject A voice recording apparatus comprising: a camera unit for picking up an image of the object; and a microphone for picking up the ambient sound, the microphone other than a power binaural microphone using a binaural microphone mounted on the ear of the photographer is switched A switching unit, a video processing unit for processing the video signal outputted from the force melting unit, an audio processing unit for processing a sound signal outputted from a microphone for collecting the ambient sound, And when switching to the binaural microphone as the ambient sound, the sound mode for collecting the ambient sound is set as a binaural mode, A flag information generating section for generating a binaural flag signal indicating binaural mode, a flag information generating section for generating a binaural flag signal indicating binaural mode, a flag information generating section for generating a binaural flag signal indicating binaural mode, And a recording unit for recording the video and audio data on the recording medium.

[0009] According to the present invention, it is possible to reproduce a sound with realistic sensation together with a captured image irrespective of the size and shape of the apparatus main body. In addition, when photographing by zooming up the subject, it is possible to reproduce sound with more realistic interlocking with the zoom-up of the subject. Furthermore, even in the case where an arbitrary photographer shoots and watches an arbitrary viewer, as in the case where the viewer viewing the reproduction signal and the photographer who captures the binaural are different from each other, there is almost no sense of incompatibility I can listen to a sound with a sense

In this configuration, as the built-in microphone built in the video and audio recording apparatus, an external microphone connection terminal, and an external microphone connected to the external microphone connection terminal, a microphone other than the binaural microphone power binaural microphone , A connection detection section for detecting whether or not an external microphone is connected to the external microphone connection terminal, and a connection detection section for outputting, as an audio signal to be supplied to the audio processing section, A switch for switching between an audio signal output from said external microphone and an audio signal output from said external microphone, and said external microphone is set as a binaural microphone by said setting unit, and an external microphone is connected to said external microphone connection terminal by said connection detection unit And a control unit for setting the audio mode to the binaural mode when it is detected that it is connected Wherein the control unit performs switching control of the switch so that the switch supplies the audio signal output from the external microphone to the audio processing unit when the audio mode is the binaural mode, It is preferable that the generation unit controls the flag information generation unit to generate the binaural flag signal.

[0010] The display control unit includes a display unit for displaying the video signal output from the camera unit, and a display control unit for displaying a binaural mark indicating that the binaural mode is displayed on the display unit when the audio mode is the binaural mode .

[0011] Furthermore, the camera section has a zoom function of enlarging and capturing the subject, and includes a sound zoom processing section that amplifies the sound signal collected by the binalolic microphone according to the enlargement ratio in the camera section It is preferable.

[0012] Furthermore, the camera unit has a zoom function for enlarging and capturing the object, and the sound source of the sound signal collected by the binaural microphone is virtually brought close to the listener to form a virtual sound source A transfer function memory for storing a head related transfer function of the head sound source in association with a plurality of distances between the virtual sound source and the listener and a plurality of head related transfer functions stored in the transfer function memory, A convolution arithmetic unit for convoluting the head related transfer functions acquired by the function acquiring unit with respect to the sound signals collected by the binaural microphone, And a sound voice zoom processing unit having a sound processing unit.

[0013] In order to solve the problem of the above-described conventional technique, the present invention provides an image recording apparatus for recording an image signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer including a sound emitted from the subject An audio recording method comprising: an image capturing step of capturing an image of the object; and switching

whether to use a microphone other than a power binaural microphone that uses a binaural microphone to be worn by the photographer's ear as a microphone that collects the ambient sound A video processing step of processing a video signal obtained by imaging the subject; an audio processing step of processing an audio signal outputted from a microphone that collects the ambient sound; When switching to the binaural microphone as a microphone for collecting the ambient sound, A flag information generating step of generating a binaural flag signal indicating that the binaural mode is set as a binary mode as a binary mode; And a recording step of recording the image signal and the noisy ralph lag signal on a recording medium.

[0014] In order to solve the problem of the above-described conventional technique, the present invention is characterized in that a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of the photographer including the sound emitted from the subject force are recorded A reproducing unit for reproducing a recording signal recorded on the recording medium by a video / audio reproducing device for reproducing a recording medium; and a recording unit for reproducing the recording signal reproduced by the reproducing unit on the basis of the video signal and the audio signal A video processing unit for processing the video signal separated by the separating unit, an audio processing unit for processing the audio signal separated by the separating unit, When a binaural flag signal indicating that a binaural microphone to be worn by the photographer's ear is used as the microphone for collecting the ambient sound is recorded on the recording medium, the binaural flag A flag information acquiring unit that acquires a lag signal from the speaker, a crosstalk signal that is generated when the flag information acquiring unit acquires the binaural flag signal, when the sound signal processed by the audio processing unit is sounded by a speaker And a crosstalk canceller for processing the audio signal so as to cancel the speech signal, wherein the crosstalk canceller has a filter convolving filter characteristics obtained in advance with respect to the speech signal, Wherein the filter characteristic is a filter characteristic based on a head related transfer function measured using a sound signal collected by a pair of microphones mounted on the structure.

[0015] A video and audio reproduction method for reproducing a recording medium on which a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer including a sound emitted from the subject are recorded, A separating step of separating the video signal and the audio signal from the recording signal reproduced in the reproducing step; a separating step of separating the video signal and the audio signal from the recording signal reproduced in the reproducing step; An audio processing step of processing the audio signal separated by the separating step; an audio processing step of processing the audio signal separated by the separating step as a microphone for collecting the ambient sound on the recording medium; When a binaural flag signal indicating that a binaural microphone to be mounted on the camera is used is recorded, the binaural flag signal is taken A flag information acquisition step of acquiring a binaural flag signal at the flag information acquisition step, canceling a crosstalk signal generated when the sound signal processed by the audio processing step is sounded by a speaker And a crosstalk cancellation step of processing the audio signal so as to process the audio signal, wherein the crosstalk cancellation step is a step of convolving a filter characteristic obtained in advance with respect to the audio signal, Characterized in that the filter characteristic is a filter characteristic based on a head related transfer function measured by using a sound signal collected by a pair of microphones mounted on a type structure.

Brief Description of the Drawings

[0016] FIG. 1 is an external perspective view showing a video / audio recording / reproducing apparatus according to a first embodiment of the present invention.

FIG. 2 is a diagram showing a state in which a subject is photographed using the video / audio recording / reproducing apparatus according to the first embodiment of the present invention.

FIG. 3 is a block diagram showing an internal configuration example of a video / audio recording / reproducing apparatus according to a first embodiment of the present invention.

FIG. 4 is a diagram showing a display screen for performing initial setting of audio mode in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 5 is a diagram showing a display example of a binaural rakemake in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 6 is a diagram showing a modified example of the binaural rakemake used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 7 is a diagram showing a modified example of binaural rakemake used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

[FIG. 8] FIG. 8 is a diagram showing a modified example of a binaural mike used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 9 is a diagram showing an example of a description format of a binaural Ralph Lag signal in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

[FIG. 10] FIG. 10 is a diagram showing another example of the description format of the binauraflag signal in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

[FIG. 11] FIG. 11 is a diagram showing still another example of the description format of the binauraflag lag signal in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 12 is a flowchart for explaining a recording operation in the video / audio recording / reproducing apparatus according to the first embodiment of the present invention.

FIG. 13 is a flowchart for explaining a playback operation in the video / audio recording / playback apparatus according to the first embodiment of the present invention.

[FIG. 14] FIG. 14 is a block diagram showing a configuration example of a crosstalk canceler used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

[FIG. 15] FIG. 15 is a diagram showing a head related transfer function measuring device for obtaining a head related transfer function characteristic used in a crosstalk canceler used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 16 is a view showing a cylindrical structure to which a microphone unit used in the head related transfer function measuring apparatus shown in FIG. 15 is attached and a dummy head microphone for comparison. 17] FIG. 17 is a waveform diagram showing an impulse response waveform measured by the head related transfer function measuring apparatus shown in FIG. 15.

(18) FIG. 18 is a waveform diagram showing frequency characteristics measured by the head related transfer function measuring apparatus shown in FIG.

FIG. 19 is a waveform diagram showing an impulse response waveform measured using a dummy head microphone.

FIG. 20 is a waveform diagram showing frequency characteristics measured using a dummy head microphone

FIG. 21 is a diagram for explaining a crosstalk cancellation characteristic based on a filter characteristic based on a head related transfer function measured using a cylindrical structure having a microphone unit attached thereto.

FIG. 22 is a diagram for explaining crosstalk canceling characteristics by filter characteristics based on head related transfer functions measured using a dummy head microphone.

FIG. 23 is a diagram for explaining crosstalk canceling characteristics by filter characteristics based on a head related transfer function measured using a cylindrical structure having a microphone unit attached thereto.

[FIG. 24] FIG. 24 is a diagram for explaining crosstalk cancellation characteristics by filter characteristics based on head related transfer functions measured using a dummy head microphone.

FIG. 25 is a block diagram showing another configuration example of a crosstalk canceler used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

[FIG. 26] FIG. 26 is a block diagram showing still another configuration example of the crosstalk canceler used in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 27 is a flowchart showing a playback operation on a headphone in the video / audio recording / reproducing apparatus according to each embodiment of the present invention.

FIG. 28 is a block diagram showing an internal configuration example of a video / audio recording / reproducing apparatus according to a second embodiment of the present invention.

FIG. 29 is a block diagram showing a configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the second embodiment of the present invention. [FIG. 30] FIG. 30 is a flowchart for explaining the operation in the case of performing the voice zoom processing in the video / audio recording / reproducing apparatus according to the second embodiment of the present invention.

[FIG. 31] FIG. 31 is a block diagram showing another configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the second embodiment of the present invention.

32 is a diagram showing a head related transfer function measuring device for finding a head related transfer function used in another configuration example of the audio zoom processing unit in FIG. 31.

FIG. 33 is a cross-sectional view showing a dummy head microphone used in the head related transfer function measuring apparatus of FIG. 32.

[FIG. 34] FIG. 34 is a diagram showing the characteristics of a head related transfer function obtained by measuring with the head-related transfer function measuring apparatus of FIG. 32.

FIG. 35 is a diagram showing characteristics of a head related transfer function obtained by measuring with the head-related transfer function measuring apparatus of FIG. 32.

36 is a diagram showing the characteristics of the head-related transfer function obtained by measuring with the head-related transfer function measuring apparatus of FIG. 32. FIG.

37 is a diagram showing characteristics of a head-related transfer function obtained by measuring with the head-related transfer function measuring apparatus of FIG. 32. FIG.

38 is a diagram showing characteristics of a head related transfer function obtained by measuring with the head related transfer function measuring apparatus of FIG. 32.

39 is a diagram showing the characteristics of the head-related transfer function obtained by measuring with the head-related transfer function measuring apparatus of FIG. 32. FIG.

FIG. 40 is a block diagram showing the operation of the audio-video recording / reproducing apparatus according to the second embodiment of the present invention in the case of performing audio zoom processing using another configuration example of the audio zoom processing section shown in FIG. 31 7 is a flow chart for explaining.

[FIG. 41] FIG. 41 is a block diagram showing an internal configuration example of a video / audio recording / reproducing apparatus according to a third embodiment of the present invention.

FIG. 42 is a block diagram showing a configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the third embodiment of the present invention.

FIG. 43 is a block diagram showing another configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the third embodiment of the present invention.

FIG. 44 is a block diagram showing an internal configuration example of the video / audio recording / reproducing apparatus according to the fourth embodiment of the present invention.

FIG. 45 is a block diagram showing a configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the fourth embodiment of the present invention.

FIG. 46 is a flowchart for explaining the audio sound zoom process in the video / audio recording / reproducing apparatus according to the fourth embodiment of the present invention.

FIG. 47 is a block diagram showing an internal configuration example of a video / audio recording / reproducing apparatus according to a fifth embodiment of the present invention.

[FIG. 48] FIG. 48 is a block diagram showing a configuration example of a voice zoom processing unit in the video / audio recording / reproducing apparatus according to the fifth embodiment of the present invention.

49 is an external perspective view showing a video / audio recording / reproducing apparatus according to a sixth embodiment of the present invention. FIG.

FIG. 50 is a block diagram showing an internal configuration example of a video / audio recording / reproducing apparatus according to a sixth embodiment of the present invention.

51 is a plan view showing an example of the structure of a cord housing section in a video / audio recording / reproducing apparatus according to a sixth embodiment of the present invention. FIG.

FIG. 52 is an external perspective view showing a video / audio recording / reproducing apparatus according to a seventh embodiment of the present invention.

FIG. 53 is a block diagram showing an internal configuration example of the video / audio recording / reproducing apparatus according to the seventh embodiment of the present invention.

[FIG. 54] FIG. 54 is a block diagram showing a specific configuration example of a wireless binaural microphone and a radio transmission / reception unit in the video / audio recording / reproducing apparatus according to the seventh embodiment of the present invention.

FIG. 55 is a diagram for explaining a warning when the wireless binaural microphone crosses the communicable area in the video / audio recording / reproducing apparatus according to the seventh embodiment of the present invention.

56 is a view showing an example of a warning mark displayed on the display unit when the wireless binaural microphone exceeds the communicable area in the video / audio recording / reproducing apparatus according to the seventh embodiment of the present invention .

57 is a flowchart for explaining the operation of the video / audio recording / reproducing apparatus according to the seventh embodiment of the present invention.

BEST MODE FOR CARRYING OUT THE INVENTION

[0017] DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS Hereinafter, embodiments of the video and audio recording apparatus and method, the video and audio reproducing apparatus and method of the present invention will be described with reference to the drawings.

[0018] ⟨⟨ First embodiment ⟩⟩

FIG. 1 is a perspective view showing an external configuration example of the video / audio recording / reproducing apparatus 101 according to the first embodiment.

[0019] The video / audio recording / reproducing apparatus 101 shown in FIG. 1 includes a camera section 11, a display section 17, internal stereo microphones 21 a and 21 b, and an external microphone connection terminal 32. An earphone type binaural microphone 3 having right and left microphones 31a and 31b of omnidirectionality is detachably connected to the external microphone connection terminal 32. [Here, a state where the binaural microphone 3 is connected to the external microphone connection terminal 32 is illustrated. The microphones 31a and 31b have diaphragms therein. As will be described in detail later, the video / audio recording / reproducing apparatus 101 selectively performs photography (sound recording) using the built-in

stereo microphones 21 a and 21 b and photography (binaural recording) using the binaural microphone 3 . Note that shooting means not only capturing an image of a subject but also capturing the image of the subject and collecting the ambient sound of the photographer including the sound emitted from the subject May occur.

[0020] FIG. 2 shows a state where the photographer 300 shoots a subject (not shown) using the video / audio recording / reproducing apparatus 101. When taking a subject while collecting sound using the binaural microphone 3, as shown in FIG. 2, the photographer 300 attaches left and right microphones 31 a and 31 b to the left and right ears 302. As a result, the ambient sound of the photographer 300 including the sound emitted from the subject with the binaural acoustic characteristics given by the positional relationship between the head 30 of the photographer 300 and the microphones 31a and 31b is collected. While viewing the monitor image of the object displayed on the display unit 17, the photographer 300 photographs the subject with the camera unit 11 and collects the surrounding sound with the binaural microphone 3. As will be described in detail later, the video signal picked up by the camera section 11 and the audio signal picked up by the binaural microphone 3 are recorded on a recording medium, not shown here. Also, as will be described in detail later, the video signal recorded on the recording medium is reproduced together with a voice having a realistic feeling as if the viewer is viewing in the same shooting environment as the photographer 300.

[0021] FIG. 3 is a block diagram showing a concrete internal configuration example of the video / audio recording / reproducing apparatus 101.

[0022] The video / audio recording / reproducing apparatus 101 includes a camera unit 11, a video encoding unit 12, a multiplexing unit 13, a recording / reproducing unit 14, a demultiplexing unit 15, a video decoding unit 16, a display unit 17, a built-in stereo microphone 21 An audio decoding unit 26, a crosstalk canceller 27, an external microphone connection terminal 32, a flag information acquisition unit 36, a video output terminal 37a, an audio output terminal 37b, a connection A detection unit 41, a flag information generation unit 42, a recording medium 44, a control unit 47, an operation unit 48, and switches Sw 1, Sw 2, Sw 3. The recording medium 44 may be a removable recording medium such as a disc-like recording medium or a tape cassette, and may be a recording medium such as a hard disc that is mounted in the video / audio recording / reproducing apparatus 101 in advance and is V Even if it is there! /,.

[0023] A monitor 52 such as a television receiver is connected to the video output terminal 37 a. Speakers 53 and 54 are connected to the audio output terminal 37 b via an amplifier 51. The sound produced by the speakers 53 and 54 is listened to by the viewer 59. In FIG. 3, for the sake of convenience, for the sake of convenience, it is shown that the photographer 300 and the viewer 59 are illustrated simultaneously. Normally, photographing by the photographer 300 and watching of the reproduced image / sound by the viewer 59 are separately performed Yes.

[0024] ⟨Recording Operation⟩

The recording operation in the video / audio recording / reproducing apparatus 101 will be described.

[0025] First, when the photographer 300 performs an operation to display the initial setting image (window) of the sound mode by the operation unit 48, the control unit 47 displays the initial setting image 170 as shown in FIG. 4 on the display unit 17 as an example . Both the binaural microphone 3 as described in FIGS. 1 and 2 and a normal external microphone can be connected to the external microphone connection terminal 32 as an external microphone. When the photographer 300 collects sounds using the noisel microphone 3 by a predetermined operation by the operation unit 48, it selects "binaural" as shown in FIG. 4, and collects sounds using a normal external microphone Select "Normal" if you want. The control unit 47 serves as a setting unit for setting as an external microphone connected to the external microphone connection terminal 32 whether to use a microphone other than the innovative microphone 3 or the binaural microphone. When "binaural" is selected as the external microphone input and the connection detecting section 41 detects that the plug of the external microphone is inserted in the external microphone connecting terminal 32, the control section 47 controls the video / audio recording / reproducing apparatus 101 controls each part of the circuit so as to perform a recording operation corresponding to photographing using the binaural microphone 3. The sound mode in which the binaural microphone 3 is used to sound ambient sounds and the collected sound signals are recorded is referred to as binaural mode. The ambient sound is collected using the built-in stereo microphone 21 or an ordinary external microphone, and the sound mode in which the collected sound signal is recorded is set to the normal mode.

[0026] If the plug of the binaural microphone 3 is formed exclusively for the binaural microphone 3 different from the usual external microphone and the external microphone connection terminal 32 is used as a connection terminal dedicated to the binaural microphone 3, the initial setting of the above-mentioned sound mode can be omitted can do.

[0027] Specifically, in FIG. 3, when detecting that the external microphone is connected to the external microphone connection terminal 32, the connection detection unit 41 supplies a detection signal to the control unit 47. When a detection signal indicating that an external microphone is connected is input while the external microphone is set to "binaural", the control unit 47 switches the switch Swl to a terminal a for inputting a sound signal from the built-in stereo microphone 21 To the terminal b for inputting the audio signal from the binaural microphone 3. As a result, the audio signal from the binaural microphone 3 is supplied to the audio coding section 22. The switch Swl serves as a switching unit for switching whether to use a microphone other than a force binaural microphone that uses a no noise mic to be attached to the ear 30 2 of the photographer 300 as a microphone for collecting ambient sounds.

[0028] In addition, the control unit 47 controls to generate and output flag information (binaural flag signal) indicating that the flag information generation unit 42 is in binaural mode. The binaural flag signal is input to the multiplexing unit 13.

[0029] When the binaural mode is set in this manner, the control unit 47 preferably displays a mark indicating that the binaural mode is set on the display unit 17. An example of the mark is shown in FIG. A mark 171 shown in FIG. 5A is a mark schematically showing the photographer 300 using the binaural microphone 3, and a mark 172 shown in FIG. 5B schematically shows a sphygmomanic reproduction of binaural sound It is a mark shown. As the mark indicating the binaural mode, any of the marks in FIGS. 5A and 5B may be used, but other marks may be used. If these marks are displayed on the display section 17 after being superimposed on the image photographed by the camera section 11 after the above-described initial setting or when the binaural microphone 3 is connected to the external microphone connection terminal 32 Yo With such a mark display, the photographer 300 can confirm whether the audio mode is the force set to the binaural mode when using the binaural microphone 3. When the audio mode is the binaural mode, the control unit 47 is a display control unit for displaying a binaural mark (marks 171, 172) indicating that it is in the binaural mode on the display unit 17.

[0030] The viewer 300 wearing the left and right microphones 3 la, 3 lb of the binaural microphone 3 on the left and right ears 302 images the subject with the camera unit 11. The video signal output from the camera unit 11 is input to the video encoding unit (video processing unit) 12 and the terminal g of the switch Sw 3. When photographing (recording) is being performed by the video / audio recording / reproducing apparatus 101, the switch Sw 3 is switched to the terminal g, the video signal from the camera section 11 is supplied to the display section 17, An image is displayed. At the same time, due to the positional relationship between the head 30 of the photographer 300 and the microphones 31a, 31b, the microphone 31a, 31 obtains a sound signal with a noisy sound whose subject is in the median plane direction. This audio signal is input to the audio coding section (audio processing section) 22 via the switch Swl.

[0031] In the case where the recording medium 44 is a DV cassette, the video encoding section 12 AZD converts the input video signal and encodes it according to the DV compression method to generate an encoded video signal. The audio coding section 22 AZD converts the input audio signal and rearranges the data positions of the uncompressed audio signal by shuffling to generate a coded audio signal.

[0032] The multiplexing unit 13 time division multiplexes the input coded image signal, the coded voice signal, and the binaural flag signal according to the signal format defined by the consumer digital VCR specification to generate a multiplexed signal . The multiplexed signal output from the multiplexing unit 13 is input to the recording / reproducing unit 14. The recording / reproducing unit 14 records the input multiplexed signal on the recording medium 44 in accordance with the recording format defined by the consumer digital VCR specification. The specific recording method of the Binotto-Ralf-Lag signal will be described in detail later. [0033] Here, a modified example of the binaural microphone 3 will be described.

[0034] ⟨Modified Example of Binaural Microphone 3⟩

FIG. 6 (A) shows a microphone 31 c as a first modification of the microphones 31 a and 31 b, and FIG. 6 (B) shows a microphone 31 d as a second modification of the microphones 31 a and 31 b. A microphone 31 c shown in FIG. 6 (A) is provided to be connected to an upper part of the microphone holding part 312 and a microphone holding part 312 to be inserted into the ear 302 of the photographer 300, and houses a microphone unit such as a

diaphragm And a microphone housing part 311 for making it! By making the binaural microphone 3 a microphone 31 c structured to be separated into the microphone storage part 311 and the microphone holding part 312, even if the photographer 300 mounts the binaural microphone 3, it is possible to listen to the sound arriving from the outside satisfactorily it can. The microphone 31 d shown in FIG. 6 (B) is provided with a microphone storage part 311 connected to the lower part of the microphone holding part 312. Even microphone 3 Id has the same effect as microphone 31 c.

[0035] FIG. 7 is a perspective view showing a specific configuration example of the microphone holding unit 312. Here, the microphone 31 c of FIG. 6 (A) in which the microphone housing portion 311 is provided on the top of the microphone holding portion 312 is shown. The microphone holding portion 312 shown in FIG. 7 (A) has a holding portion main body 312 a formed with a tapered sound passage hole 313 a whose diameter decreases toward the inside of the ear 302. The microphone holding portion 312 shown in FIG. 7 (B) is formed by forming the holding portion main body 312 a circular columnar sound hole 313 b. The holder main body 312a of FIG. 7 (A) is easy to insert into the ear 302 of the photographer 300, and the holding part main body 312b of FIG. 7 (B) has the feature that the attenuation of external sound when worn is small.

[0036] FIG. 8 shows an example in which the external shape of the microphone holding part 312 of the microphone 31 c in FIG. 6 (A) is made different. 8A shows a microphone holding portion 312 having a large outer shape, FIG. 8B shows a microphone holding portion 312 having an outer shape, and FIG. 8C shows a microphone holding portion 312 having a small outer shape. In this way, if a plurality of sizes are prepared as the microphone holding unit 312, the photographer 300 can select one matching the size of his ear 302. 8A to 8C, the shape and size of the microphone housing portion 311 (microphone unit) are the same, and the sensitivity and response frequency characteristics of the microphone unit are also the same.

[0037] ⟨Recording of binaural flag signal⟩

In order to distinguish between the binaural sound collected by the photographer 300 wearing the binaural microphone 3 and the stereo sound collected by the built-in stereo microphone 21, binaural sound and a binaural sound signal are recorded on the recording medium 44 at the time of binaural sound collection Record. The binno-lull-lag signal is generated by the flag information generating unit 42.

[0038] First, a specific recording method of the binaural flag signal when the recording medium 44 is a DV cassette will be described.

[0039] FIG. 9 shows a data format for recording audio data in the DV cassette.

0When recording audio data of ~ 89 bytes, the 0th and 1st bytes are synchronous codes, the 2nd to 4th by ID (identification) code, the 5th to 9th bytes are audio auxiliary data (AUX) 10 to 81 record audio data, and 82 to 89 record inner code parity data for error data detection and correction. For example, the flag information generating unit 42 outputs 1 as a binaural flag signal in binaural mode, and outputs 0 as long as it is not in binaural mode (in normal mode). The multiplexing unit 13 generates the data format signal shown in FIG.

[0040] Next, a specific recording method of the binaural flag signal in the case where the recording medium 44 is a disk for recording will be described. As a disk for recording, there are a disk for recording and reproducing using red laser light such as DVD-RAM, DVD-RW, DVD-R and a blue laser light such as Blu-ray Disc and HD-DVD There is a disk to record and reproduce. Here we will explain the case of superimposing the binauraflag lag signal using the DVD video standard that can be applied to these recording discs in general.

[0041] As a first method of superimposing the Noino-Ralph lag signal using the DVD video standard, a method of superimposing a binaural flag signal on a DVD-Video zone in the DVD video standard can be considered.

[0042] As shown in Fig. 10, the Volume space of the DVD standard is powerful with Volume and Film structure, DVD - Video zone and DVD others zone. The DVD-Video zone includes V MG (Video Manager) and VTS (Video Title Set) # 1 to DVTS # n. n is a predetermined integer of 2 or more. Each VTS includes control data (Control Data) and VOBS (Video Object Set). The VOBS includes a plurality of VOBs (Video Objects). The VOBS includes a plurality of CELLs. CELL includes a plurality of VOBU (Video Object Unit). VOBU includes navigation pack (NV_PACK), audio pack (A_PACK), video pack (V_PAC K). In the present embodiment, a data pack (D_PACK) including a binaural flag signal is provided in the VOBU.

[0043] The data pack (D_PACK) includes pack header, packet header, substream ID, audio frame information, audio data information, and noino ralph lag signal. Noino Ralph Lag signal will also be multiple audio frame layer power.

[0044] In this manner, information including a binaural flag signal in a format conforming to the DVD video standard is packed as a data pack (D_PACK) and MPEG multiplexed. This makes it possible to know which audio frame portion of the audio signal is a binalaural audio signal and which audio frame portion is a normal stereo sound while maintaining compatibility with the DVD video standard. Therefore, it is possible to easily determine in which audio frame portion the crosstalk cancellation process should be performed.

[0045] As a second method of superimposing the Noino-Ralph lag signal using the DVD video standard, a method of superimposing a binaural flag signal on DVD other zone in the DVD video standard can be considered. The DVD other zone is an area for recording auxiliary data related to data of the main body of video and audio, and it is also a user data recording area.

[0046] In the present embodiment, the data structure of the DVD other zone which is the user data recording area is structured similar to the data structure of the DVD-Video zone so as to be apportioned by comparing FIG. 10 with FIG. 11. As shown in FIG. 11, the DVD other zone also includes VMG, VTS, VOBS, VOB, CELL, and VOBU. In order to distinguish these pieces of information in the DVD other zone shown in FIG. 11 from that of FIG. 10, the name given with D is added to the head.

[0047] As shown in FIG. 11, the DVD other zone includes DVMG and DVTS # 1 to DVTS # n.

Each DVTS includes DVTSI (Video Title Set Information) and DVOBS. The DV OBS includes a plurality of DVOBs. DVOBS contains multiple DCELLs. DCELL includes a plurality of DVOBUs. The DVOBU includes a plurality of audio frame layers. The audio frame layer is an area for recording audio frame data such as coding parameters of audio signals, and a part of this audio frame layer is set as the recording area of the binaural flag signal.

[0048] By describing the binaural flag signal in the DVD other zone in the format conforming to the DVD video standard as described above, audio signals (binaural audio signal or normal stereo sound) included in the DVD-Video zone and binaural flag signal Are associated with each other. It is possible to know which audio frame portion of the audio signal is a binaural sound signal and which audio frame portion is a normal stereo sound while maintaining compatibility with the DVD video standard. Therefore, it is possible to easily determine in which audio frame portion the crosstalk cancellation process should be performed.

[0049] In the cases of FIGS. 10 and 11, for example, an audio signal picked up while photographing is started by operating the start button and the stop button is operated to end shooting is set as one or a plurality of audio frame layers, Add audio mode information to each audio frame layer. Then, the binaural flag signal is described in the voice mode information and managed as a binaural information packet. By managing the Noino Ralph lag signal as a binaural information packet, it is possible to easily acquire voice mode information for each audio frame. Thus, even when the binaural audio signal and the normal stereo audio signal are mixedly recorded at the time of reproducing the recording medium 44, the crosstalk canceller 27 Can be switched appropriately. Recording of the audio mode information is required to be performed at least for each operation of starting photographing, and it is more preferable to record at every predetermined time.

[0050] Also, even in the case of using, for example, a semiconductor memory as the recording medium 44, by defining the recording area of the binaural flag signal and specifying the sound mode of the sound signal to be recorded in the same manner as described above, it is possible to record binaural It is possible to identify the audio signal and reproduce the audio signal while appropriately switching the operation / non-operation of the crosstalk canceler 27.

[0051] It is also conceivable to describe the binaural flag signal in the user data of the multiplexing layer of the encoding method such as MPEG. As an example, it is assumed that a cellular phone having a communication function of video and audio is used, and a case where a video signal photographed together with a voice signal collected by the binaural microphone 3 from a cellular phone on the transmitting side is transmitted to the mobile phone on the receiving side . Even in such a case, the binaural flag signal can be transmitted from the transmitting-side cellular phone to the receiving-side cellular phone. By transmitting the binaural flag signal to the audio signal and transmitting it, it becomes possible to reproduce a sound with realism in the case of binaural sound. In this case, a binaural flag signal is described in a predetermined portion of the video and audio

packet data to be transmitted by the cellular phone and transmitted as packet data. For example, when using the transmission method specified by MPEG-4, it is possible to transmit a no-no-ruff-lag signal as shown in FIG. 9 in the user data recording area of the elementary stream. When a transport stream specified by the MPEG -4 standard is used, a binaural flag signal can be described in a private data area (private_data_type).

[0052] Also, when transmitting video data and audio data as fileed data in the form of attached files, binaural flag signals are written in the file header portion.

[0053] The recording operation in the video / audio recording / reproducing apparatus 101 will be described further using the flow chart shown in FIG.

[0054] First, in step S151, the control unit 47 determines whether the external microphone connected to the initial setting force external microphone connection terminal 32 described in FIG. 4 is set in the binaural microphone 3 or not. If it is determined in step S 151 that the initial setting is binaural (YES), the process proceeds to step S 152. If it is not determined that the binaural setting is made (NO), the control unit 47 switches the switch Swl to the terminal a. In step S154, the video / audio recording / reproducing apparatus 101 transmits the audio signal from the built-in stereo microphone 21 get. In step S152, the control unit 47 determines whether or not the connection detecting unit 41 is detecting the insertion of the plug of the external microphone into the external microphone connecting terminal 32. [In step S 152, if it is determined that an external microphone is connected to the external microphone connection terminal 32 (YES), the control unit 47 switches the switch Swl to the terminal b, and in step S153, the video / audio recording / reproducing apparatus 101 acquires an audio signal from the no-noal microphone 3. If it is not determined in step S 152 that an external microphone is connected to the external microphone connection terminal 32 (NO), the control unit 47 switches the switch Swl to the terminal a, and in step S 154, The reproduction apparatus 101 acquires a sound signal from the built-in stereo microphone 21.

[0055] Next, in step S155, the video signal from the camera section 11 is temporarily stored in a memory (not shown) in the video encoding section 12, and the audio signal from the binaural microphone 3 or the built-in stereo microphone 21 is temporarily stored in the audio encoding section In a memory (not shown) in the unit 22. At step SI 56, the video encoder 12 encodes the video signal, and the audio encoder 22 encodes the audio signal. In step S157, the coded video signal is temporarily stored in a buffer (not shown) in the video encoding section 12, and the coded audio signal is temporarily stored in a buffer (not shown) in the audio coding section 22. Then, Further, in step S158, the flag information generating unit 42 generates a binaural flag signal based on an instruction from the control unit 47 in the binaural mode.

[0056] Then, in step S159, the multiplexing unit 13 multiplexes the encoded video signal, the encoded audio signal, and the binaural flag signal, and in step S 160, generates the packetized stream signal. In step S 161, the recording / reproducing unit 14 records a stream signal in the medium recording 44. In step S162, the video encoder 12 and the audio encoder 22 determine whether there is a video signal and a sound signal to be encoded, and it is determined that there is still a video signal and a sound signal to be encoded (YES), the process returns to step S152, and the above operation is repeated. If it is not determined in step S162 that there is still a video signal and an audio signal to be coded (No), the process is terminated.

[0057] ⟨Playback Operation⟩

Returning to FIG. 3, the playback operation in the video / audio recording / playback apparatus 101 will be described. In FIG. 3, when a reproduction button (not shown) of the operation unit 48 is operated, the recording / reproduction unit 14 reproduces the multiplexed signal which is the recording signal recorded on the recording medium 44 under the control of the control unit 47. The multiplexed signal reproduced by the recording and reproducing unit 14 is supplied to a demultiplexing unit 15 which separates the input multiplexed signal into a coded video signal, .

[0058] The coded video signal is supplied to a video decoding section (video processing section) 16, the coded audio signal is supplied to an audio decoding section (audio processing section) 26, and the noino ralphag signal is transmitted as flag information And is supplied to the acquisition unit 36. The video decoding section 16 decodes the input coded image signal to obtain a video signal. In accordance with the operation of the reproduction button, the switch Sw 3 is switched to the terminal h under the control of the control unit 47. The video signal outputted from the video decoding section 16 is displayed on the display section 17 and also supplied to the monitor 52 via the video output terminal 37a for display. The audio decoding section 26 decodes

the input coded voice signal to obtain a voice signal. The audio signal is supplied to the crosstalk canceller 27 and the terminal c of the switch Sw 2.

[0059] The crosstalk canceler 27 generates a first tone signal that is generated when a binaural collected sound signal is reproduced by the speakers 53 and 54 and a signal sounded from the left speaker 54 is listened to by the right ear of the listener 59 Generates a signal for canceling the crosstalk component and the second crosstalk component that the signal generated by the right speaker 53 is listened to by the left ear of the listener 59 and adds it to the audio signal And generates a crosstalk processed signal. The flag information acquisition unit 36 holds the binaural flag signal obtained by the separation unit 15. The control unit 47 switches over the switch Sw 2 depending on whether the flag information acquisition unit 36 holds the binaural flag signal or not. That is, if the flag information acquiring unit 36 holds the binaural flag signal, if the switch Sw 2 is connected to the terminal d and the crosstalk canceling signal of the crosstalk canceller 27 power is also output to the sound output terminal 37 b If it is not controlled and the switch Sw 2 is connected to the terminal c and the audio signal which is not crosstalk processed from the audio decoding unit 26 is transmitted to the audio output terminal 37 b So as to output.

[0060] The sound signal output from the sound output terminal 37 b and amplified by the amplifier 51 is sounded from the left and right speech forces 53, 54. When the audio signal output from the audio output terminal 37 b is the crosstalk processed signal output from the crosstalk canceler 27, the viewer 59 views the video displayed on the monitor 52, and the photographer 300 captures an image It is possible to listen to the surrounding sound of the photographer 300 at the time of going as a sound with realism. At this time, since the crosstalk canceller 27 cancels the crosstalk component by using the head-related transfer function which will be described in detail later, even if the photographer 300 and the viewer 59 are different from each other, an arbitrary photographer 300 So that even when an arbitrary viewer 59 views it, he / she can listen to a sound with realistic feeling with little sense of discomfort.

[0061] The reproduction operation in the video / audio recording / reproducing apparatus 101 will be described further using the flow chart shown in FIG.

[0062] Referring to FIG. 13, in step S181, the recording and reproduction unit 14 reproduces the recording medium 44 and obtains a stream signal based on the multiplexed signal. In step S182, the recording and reproduction unit 14 decodes the stream signal to generate a packet signal. In step S183, the demultiplexing unit 15 separates the packet signal into a video signal, an audio signal, and a no-no-rural signal. In step S 184, the video decoding unit 16 decodes the video signal, and the audio decoding unit 26 decodes the audio signal. In step S185, the video decoding unit 16 and the audio decoding unit 26 temporarily store the decoded video signal and audio signal in a buffer (not shown). In step S186, the flag information acquiring unit 36 acquires the binaural flag signal.

[0063] Then, in step S187, on the basis of the binauraf luggage signal acquired by the flag information acquiring unit 36, the control unit 47 determines whether the reproduced audio signal is a force binaural audio signal which is a normal stereo audio signal judge. If it is determined in step S 187 that the signal is a binaural sound signal (YES), the process proceeds to step S 188. If it is determined in step S187 that the sound signal is not a binaural sound signal (NO), the process proceeds to step S 191, the control unit 47 switches the switch Sw 2 to the terminal c, and synchronizes the video signal and the audio signal .

[0064] If it is a binaural audio signal, in step S188, the control unit 47 switches the switch Sw2 to the terminal d, thereby setting the crosstalk cancellation processing by the crosstalk canceler 27 to an effective state. In step S189, the control unit 47 controls each unit of the circuit so as to synchronously reproduce the crosstalk canceled audio signal by the video signal and the crosstalk canceller 27. If it is determined in step S 190 that there is still a video signal and an audio signal to be reproduced (YES), the process returns to step S 182 and the above operation is repeated. If it is not determined that the video signal and the audio signal to be reproduced still exist in step S 190 (NO), the processing is terminated.

[0065] **〈クロストークキャンセル〉**

Here, the specific configuration and operation of the crosstalk canceler 27 will be described with reference to FIG. 14. As shown in FIG. 14, the crosstalk canceller 27 is configured to include finishers 272 a to 272 d, calo calculators 274 a and 274 b, and filters 275 a and 275 b.

[0066] In FIG. 14, the left channel signal P (t) of the binaural sound signal is filtered by the filter 2

L

72a, 272 b [This signal, right channel signal P (t) voice letter 272 c, 272 d 【This will be done. The filters 272a to 272d are sequentially assigned to the head related transfer functions h (t), h (t), h (t), h (t)

The filter characteristic (filter coefficient) created on the basis of rs lo ro is stored. The filters 272 a and 272 d have filter characteristics equivalent to the head related transfer functions h (t) and h (t), and the filters 272 b and 2

rs Is

72c has filter characteristics equivalent to those obtained by inverting the head related transfer functions h (t) and h (t). Feces

lo ro

For convenience, the filter characteristics of the filters 272a to 272d are expressed as h (t), - h (t), - h (t), h (t)

Write rs lo ro Is. Filters 272 a to 272 d are configured to add input signals P (t) and P (t)

L R

And outputs it.

[0067] The power calculator 274 a calculates the signal output from the filters 272 a and 272 c, and the filter 275 a gives a filter characteristic of d (t) to the added signal. Calo calculator 274 b adds the signals output from filters 272 b and 27 2 d, and filter 275 b gives a filter characteristic of d (t) to the added signal.

[0068] Here, the filter characteristic d (t) stored in the filters 275 a and 275 b is the following equation (1).

[0069] d (t) = {h (t) X h (t) - h (t) X h (t)} _ 1,

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The signals output from the filters 275a and 275b are crosstalk processed signals subjected to the crosstalk cancellation processing so that the crosstalk is canceled when sound is generated from the speakers 53 and. The crosstalk processed signal output from the filters 275 a and 275 b is amplified by the left channel amplifier 5 la and the right channel amplifier 5 lb of the amplifier 51, respectively, and is sounded from the speech forces 53 and 54.

[0070] A signal (voice) generated from the speaker 53 is listened to with the left ear of the viewer 59, and a part of the sounded signal is received as a first crosstalk signal indicated by a broken line at the right ear of the viewer 59 Listen. The crosstalk canceler 27 generates a first crosstalk cancellation signal for canceling the first crosstalk signal listened to by the right ear of the viewer 59 and causes the speaker 54 to generate sound. The first crosstalk cancellation signal cancels (attenuates) the first crosstalk signal by the first crosstalk cancellation signal. Similarly, a signal (voice) generated from the speaker 54 is listened to by the right ear of the viewer 59 and a part of the sounded signal is received as a second crosstalk signal indicated by a broken line on the left side of the viewer 59 Listen with your ears. The crosstalk canceller 27 generates a second crosstalk cancellation signal for canceling the second crosstalk signal to be listened to by the left ear of the viewer 59 and causes the speaker 53 to generate sound. The second crosstalk signal is canceled (attenuated) by the second crosstalk cancellation signal. As a result, the viewer 59 listens to the audio signal P (t) whose crosstalk is canceled with the left ear, the crosstalk is canceled with the right ear

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Listens to the voice signal P (t).

[0071] ⟨Measurement of Head Related Transfer Function⟩

With reference to FIG. 15, a head related transfer function measuring device 6 for obtaining head related transfer function characteristics stored in the filters 272 a to 272 d, 275 a and 275 b will be described. As shown in FIG. 15, the head-related transfer function measuring device 6 includes a personal computer 61, an amplifier 62, a speaker 63, microphone units 65 a and 65 b, a cylindrical structure 65 e, and amplifiers 66 a and 66 b.

[0072] How to measure the head related transfer function!

[0073] First, a personal computer 61 generates a measurement signal composed of impulse sound, for example. The measurement signal is amplified by the amplifier 62. Measurement signals generated from the left speaker 63 are received by the left and right microphone units 65a and 65b. The right and left signals obtained by receiving the sound are amplified by the amplifiers 66 a and 66 b and input to the personal computer 61. This input signal is the head related transfer function h (t), h (t) of the sound received from the speaker 63 and received by the left and right microphone units 65a and 65b attached to the cylindrical structure 65e is there. Head trance

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The reaching function h (t) indicates that the signal generated from the left speaker 63 is the left microphone unit 65 a

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The head related transfer function h (t) is a characteristic to be listened to and received by the left speaker 63

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Is the characteristic of the crosstalk component listened to by the microphone unit 65 b on the right side.

[0074] Likewise, the measurement signal generated from the right speaker 64 is received by the left and right microphone units 65a and 65b. The right and left signals obtained by receiving the sound are amplified by the amplifiers 66 a and 66 b and input to the personal computer 61. The personal computer 61 compares the generated measurement signal with the received sound signal so that the sound generated from the speaker 64 is received by the left and right microphone units 65a and 65b attached to the cylindrical structure 65e Head transfer functions h (t) and h (t) are obtained. The head related transfer function h (t) is obtained from the right speaker 64

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The head related transfer function h (t) is a characteristic in which the sound generated by the right speaker 64 is a crosstalk component listened to by the left microphone unit 65a It is a characteristic.

[0075] The cylindrical structure 65 e will be described with reference to FIG. 16. 16A is a top view of the cylindrical structure 65e, FIG. 16B is a perspective view of the cylindrical structure 65e, and FIG. 16C is a cross-sectional view of a so-called dummy head microphone shown for comparison .

[0076] As shown in FIGS. 16 (A) and 16 (B), microphone units 65 a and 65 b are mounted on the surface of the cylindrical structure 65 e in a state of being separated by 180 °. As shown in the drawing, the microphone units 65a and 65b do not have an auricle and external ear canal. The diaphragm (not shown) of the microphone units 65a, 65b is disposed at a position substantially coincident with the surface of the cylindrical structure 65e. On the other hand, the dummy head microphone 69 shown in FIG. 16 (C) has pinna shaped members 692 a, 692 b and ear canals 693 a, 693 b on both sides of the artificial head 691, and a microphone unit 694 a, and 694 b, respectively. The microphone units 694 a and 694 b are provided at positions corresponding to the position of the eardrum of a person, and collect sound signals approximate to sounds to be heard by a person.

[0077] The sound receiving characteristics of the microphone units 65a and 65b attached to the cylindrical structure 65e shown in FIGS. 16 (A) and 16 (B) are influenced by characteristic differences given by auricles and external auditory canals different in size and shape from person to person And it is a characteristic that can measure the head related transfer function. Although sound waves generated from the speakers 63, 64 are shielded by the cylindrical structure 65e, they are diffracted along the cylindrical structure 65e and reach the microphone units 65a, 65b. The microphones 65a and 65b measure the characteristics of sound waves directly

emitted from the speakers 63 and 64 and reaching by diffracting along the cylindrical structure 65e. By using the cylindrical structure 65e, it is possible to obtain a head related transfer function having an average head shielding characteristic. Therefore, even when various viewers who have different sizes and shapes of the heads and have different head shielding characteristics listen to the binaural sound signal, it is possible to receive speech with realistic feeling with almost no discomfort.

[0078] 17A to 17D are diagrams showing the head related transfer function h (t (t) of the cylindrical structure 65e measured by the acoustic signal transfer characteristic measuring device 6, the impulse sound generated by the acoustic signal transfer characteristic measuring device 6 described above,), h (t), h (t), h (t) are convoluted with each other

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. FIG. 17 (D) shows the filter characteristic d (t) shown in the above formula (1). In Figs. 17 (A) - 17 (E), the vertical axis shows the amplitude of the signal voltage normalized with the prescribed output voltage, and the horizontal axis shows the number of samples when the measured signal is sampled at 48 kHz The time shown is shown.

[0079] 18 (A) to 18 (E) show the frequency characteristics obtained by Fourier analysis of the signals shown in FIGS. 17 (A) to 17 (E). In FIGS. 18 (A) to (E), the frequency positions of 100 Hz, 1 kHz, and 10 kHz are indicated by the broken vertical lines, and the gain difference between the horizontal lines of the two broken lines in the response characteristics of the vertical axis is 10 dB.

[0080] In the filters 272a to 272d of FIG. 14, the head related transfer functions h (t)

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, h (t), h (t), h (t) are given. As mentioned above, Phil lo ro Is

Filter characteristics corresponding to the head related transfer functions h (t) and h (t) are given to the tapes 272 a and 272 d

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, Filter characteristics corresponding to the head related transfer functions h (t), h (t) whose polarity is reversed - h (t) and - lo ro lo h (t) are given to the filters 272 b and 272 c.

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[0081] For comparison, the characteristics measured using the dummy head microphone 69 shown in FIG. 16 (C) instead of the microphone units 65a and 65b attached to the cylindrical structure 65e are shown in FIGS. 19 and 20. FIG. 19 shows characteristics obtained by measurement similarly to FIG. 17. As can be understood by comparing FIG. 17 with FIG. 19, the impulse response waveform measured by using the microphone units 65 a, 65 b attached to the cylindrical structure 65 e is the impulse response measured by using the dummy head microphone 69 This waveform is closer to the impulse sound which is the input signal for measurement than the waveform.

[0082] FIG. 20 shows the frequency response characteristics measured using the dummy head microphone 69. The characteristics obtained by using the microphone units 65a and 65b attached to the cylindrical structure 65e are smaller than the distortion force S of the frequency characteristic and close to flat . Respective characteristics shown in FIGS. 20 (A) to 20 (E) are enhanced or attenuated in the response characteristics at a frequency of 1.5 to 7 kHz. The characteristics shown in FIGS. 18 (A) to 18 (E) are smaller in degree of enhancement and attenuation of response characteristics. This is because there is no disturbance in characteristics caused by the auricle or the external auditory meatus when the microphone units 65a and 65b attached with the cylindrical structure 65e are used. In the dummy head microphone 69, a part of the sound waves emitted from the speakers 63, 64 is reflected by the auricle, and the reflected sound wave and the directly arriving sound wave are synthesized and synthesized in phase and synthesized in reverse phase It attenuates. In addition, it is strengthened or attenuated at a specific frequency due to the influence of external ear canal resonance, antiresonance, etc. By using the microphone units 65 a and 65 b attached to the cylindrical structure 65 e, it is possible to suppress adverse effects as in the case of using the dummy head microphone 69.

[0083] Filter characteristics based on the head related transfer function measured by using the microphone units 65a and 65b attached to the cylindrical structure 65e are added to the filters 272a to 272d and the filters 275a and 275b of the crosstalk canceler 27) And a filter characteristic based on the head-related transfer function measured using the dummy head microphone 69 (second condition), and a comparative viewing test was conducted by a plurality of viewers . A small, small microphone was inserted into the ear canal of the viewer, and the sound receiving characteristic was measured as the sound received by the small microphone is the sound heard by the viewer.

[0084] FIG. 21 shows the characteristics measured by a certain viewer under the first condition. In FIG. 21, (A) shows an impulse signal on the left side input signal P (t), a silent input signal P (t) on the right side

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This is the waveform of the impulse response signal which is sounded from the speakers 53 and 54 as the number and received by the compact microphone worn by the viewer's left ear. (B) is the waveform of the crosstalk component sounded with the same condition as (A) and received by a compact microphone worn by the viewer's right ear. In FIG. 21 (A), a large impulse response waveform is obtained, and in FIG. 21 (B), a small level waveform is obtained. FIG. 21 (C) shows the frequency analysis result of the response waveform, Ca is the response characteristic obtained by frequency analysis of the response waveform of (A), and Cb is the response characteristic obtained by frequency analysis of the response waveform of (B). Crosstalk cancellation effect exceeding 20 dB at 100 Hz to 2 kHz is obtained.

[0085] Further, in FIG. 21, (D) shows the case where the left input signal P (t) is no signal, the right input signal P

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(t) as an impulse signal from the speakers 53 and 54, and worn on the left ear of the viewer

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It is the waveform of the crosstalk component received by the small microphone. (E) is the waveform of the impulse response received by a compact microphone worn on the right ear of the viewer, which is pronounced under the same condition as (D). In FIG. 21 (D), a waveform of a small level is obtained, and in FIG. 21 (E) a large level of impulse response waveform is obtained. FIG. 21 (F) is a frequency analysis result of the response waveform, F d is a response characteristic obtained by frequency analysis of the response waveform of (D), and Fe is a response characteristic obtained by frequency analysis of the response waveform of (E) . A crosstalk cancellation effect of about 16 dB is obtained at 100 Hz to 2 kHz. FIG. 22 shows the characteristics measured by the same viewer as in FIG. 21 under the second condition. The measurement conditions are the same as in FIG. The crosstalk canceling effect of FIG. 22 (C) is about 14 dB, and the crosstalk canceling effect of FIG. 22 (F) is about l ldB. Under the second condition, the effect is inferior compared to the first condition.

[0087] Fig. 23 shows the characteristics measured with the same measurement conditions as in Fig. 21 under the first condition with different viewers from Fig. 21 and Fig. 22. The crosstalk canceling effect of FIG. 23C is about 22 dB, and the crosstalk canceling effect of FIG. 23F is about 18 dB. Even different viewers have good effects.

[0088] FIG. 24 shows the characteristics measured under the same conditions as in FIG. 22 under the second condition with the same viewers as in FIG. 23. The crosstalk canceling effect of FIG. 24 (C) is about 14 dB, and the crosstalk canceling effect of FIG. 24 (F) is about 10 dB. Even for different viewers, it can be seen that the effect is inferior to the first condition under the second condition. Furthermore, similar measurements were performed by different viewers, and it was confirmed that the same effect as above can be obtained for each of the first condition and the second condition.

[0089] From the above measurement results, the filter characteristics given to the filters 272 a - 272 d and the filters 275 a, 275 b of the crosstalk canceller 27 are more improved than the filter characteristics based on the head related transfer function measured by using the dummy head microphone 69 Direction of filter characteristics based on head related transfer functions measured using microphones 65a and 65b attached to form structure 65e Crosstalk component sounded from the left speaker and listened to by the right ear and right

side It is f * i ¾ that the effect of canceling the crosstalk component sounded by the speaker of the left ear listened to is high.

[0090] In the filter characteristics based on the head related transfer function measured using the microphone units 65 a and 65 b attached to the cylindrical structure 65 e, disturbance of the frequency characteristics in the high frequency range is small. That is, by using the cylindrical structure 65e, it is possible to reduce the frequency characteristic being greatly reduced or increased at a specific frequency portion, and it is possible to minimize the degradation of the sound quality as much as possible. As a result, the viewer can listen to the sound with realistic feeling with little sense of incongruity.

[0091] Filter characteristics based on head related transfer functions measured using the microphone units 65a and 65b attached to the cylindrical structure 65e with filter characteristics given to the filters 272a to 272d and the filters 275a and 275b of the crosstalk canceller 27 , The crosstalk cancellation is performed near the entrance of the ear canal of the viewer 59 which is the structure when listening to the binaural sound signal. Therefore, the cancellation of the crosstalk component is effectively performed for a plurality of viewers 59 having different acoustic characteristics of the auricle and the ear canal.

[0092] The cylindrical structure 65e is not limited to a complete cylindrical shape, but may be a slightly cylindrical deformed cylindrical shape! /,. It is preferable not to have a concave-convex portion that gives a change in response characteristics such as the auricle or the external auditory canal, etc. In a state where the microphone units 65 a, 65 b are attached to the cylindrical structure 65 e, the response characteristics are disturbed As much as possible, it is preferable,

[0093] The crosstalk canceler 27 is not limited to the configuration shown in FIG. 14, but may be a band division type crosstalk canceller which can further reduce the opposite phase feeling occurring in the low frequency band. The band division type crosstalk canceller divides a binaural sound signal input as a signal of the entire frequency band into a signal of a low frequency band and a signal of a mid high frequency band and converts the binaural sound signal into a binaural sound signal of a middle to high frequency band Crosstalk cancellation processing is performed only for the crosstalk.

[0094] FIG. 25 shows a band division type crosstalk canceler 27a, and its configuration and operation will be described. Constituent parts having the same functions as those of the crosstalk canceller 27 shown in FIG. 14 are denoted by the same reference numerals, and description thereof is omitted.

[0095] As shown in FIG. 25, the crosstalk canceller 27a is different from the crosstalk canceler 27 shown in FIG. 14 in that it includes low-pass filters (LPF) 271a and 271d, high-pass filters (HPF) 271b and 271c, delay units 273a and 273b, Gain control amplifiers (GC) 276 a - 276 d, calo calculators 2 77 a, 277 b Different in that they are being caught!

[0096] Among the binaural audio signals input to the crosstalk canceller 27 a, the left channel signal P (t) is input to the LPF 271 a and the HPF 271 b, the right channel signal P (t) is input to the LP

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F 271 d and HPF 271 c, and is divided into a low frequency band and a middle high frequency band. : The cut-off frequency of the LPFs 271a, 271d and the HPFs 271b, 271c is set to about 100-200 Hz. The middle and high frequency band signals output from the HPFs 271 b and 271 c are circuit portions composed of the finer switches 272 a to 272 d configured in the same manner as the crosstalk canceler 27, the calorific calculators 274 a and 274 b, and the focus switches 275 a and 275 b Crosstalk cancellation processing is performed. Signals in the middle to high frequency band subjected to the crosstalk cancellation processing are input to the gain control amplifiers 276 b and 276 c, and the gain is adjusted.

[0098] The signals in the low frequency band output from the LPFs 271a and 271d are input to the delay units 273a and 273b and are delayed by substantially the same time as the time required for the crosstalk cancellation process for the signals in the middle to high frequency band. The signals in the low frequency band output from the delay units 273 a and 273 b are input to the gain control amplifiers 276 a and 276 d, and the gain is adjusted so as to eliminate the level difference from the signals in the middle to high frequency band.

[0099] The power calculators 277 a and 277 b add the low frequency band signal output from the gain control amplifiers 276 a to 276 d and the middle high frequency band signal. The signals output from the adders 277a

and 277b are crosstalk processed signals subjected to crosstalk cancellation processing only for signals in the middle to high frequency band. The crosstalk processed signal output from the adders 277 a and 277 b is amplified by the left channel amplifier 5 la and the right channel amplifier 51 b of the amplifier 51, respectively, and is sounded from the speakers 53 and 54.

[0100] In the configuration shown in FIG. 25, crosstalk cancellation processing is not applied to signals in the low frequency band, so that it is reproduced as a signal having no antiphase feeling in the low frequency band.

[0101] In the crosstalk canceler 27 shown in FIG. 14, the crosstalk cancellation effect by the crosstalk canceller 27 is sufficiently obtained at less than or equal to 10 Hz as described above with reference to FIGS. 21 to 24. The low frequency band below lOOHz has less influence on the localization of the sound source than the V band. Without canceling the crosstalk, the signal is audibly listened to with a sense of incompatibility as a signal of opposite phase.

[0102] In the crosstalk canceller 27a shown in FIG. 25, crosstalk cancellation is not performed in a low frequency band of 100 to 200 Hz or less so as to realize a crosstalk canceller such that a reverse phase signal does not occur in a low frequency band Do it.

[0103] FIG. 26 shows a frequency band division type crosstalk canceler 27 b having a filter construction method different from that of FIG. 25, and its configuration and operation will be described. Constituent parts having the same functions as those of the crosstalk canceller 27 a shown in FIG. 25 are denoted by the same reference numerals, and description thereof is omitted.

[0104] The crosstalk canceller 27 b shown in FIG. 26 differs from the crosstalk canceler 27 a in FIG. 25 in that filters 278 a and 278 b and filters 279 a and 279 b are provided in place of the filters 272 a to 272 d and the filters 275 a and 275 b The way of wiring is also different. The crosstalk canceller 27 a is configured with a filter characteristic as a feedforward FIR (finite impulse response) type, whereas the crosstalk canceller 27 b uses a feedback type FIR type filter characteristic .

[0105] In FIG. 26, cross-track cancellation processing is performed by the FIR type filters 278 a, 278 b, 279 a, 279 b and adders 274 a, 274 b for signals in the middle to high frequency band output from the HPFs 271 b, 271 c. The filter characteristics obtained by the head-related transfer function measuring device 6 are shown in the illustration in FIG. 7 in the winding positions 278 a, 278 b, 279 a, 279 b. The winding patterns 278 a, 278 b, 279 a, 279 b The respective filter characteristics are convoluted with the input signal and output. According to the crosstalk canceller 27 b, when the discomfort is reduced as if the reverse phase signal does not occur in the low frequency band, the same action and effect as the crosstalk canceller 27 a can be obtained. The crosstalk canceller 27 b shown in FIG. 26 can reduce the order of the filter used than the crosstalk canceller 27 a shown in FIG. 25, and the configuration can be simplified. In place of FIR type filter, IIR type (infinite impulse response) infinite impulse response filter may be used.

[0106] In the configuration shown in FIG. 3, when the force control unit 47 having the crosstalk canceler 27 (or 27 a, 27 b) configured separately from the control unit 47 is a microprocessor having a DSP (Digital signal processor) , The control unit 47 may execute the functions of the crosstalk canceler 27, 27 a, 27 b. In addition, the crosstalk canceler 27, 27 a, 27 b is not limited to hardware, and can be realized by software.

[0107] ⟨Headphone playback⟩

Meanwhile, in the video / audio recording / reproducing apparatus 101 shown in FIG. 3, the audio signal outputted from the audio decoding unit 26 can be listened to even by headphones. When listening to the nooraur audio signal with headphones, the above-mentioned crosstalk components do not occur V,. When listening to the crosstalk processed signal processed by the crosstalk canceller 27 with headphones, the opposite phase component of the binaural sound signal is listened to by the left and right ears. Since the reverse phase component is an acoustic signal component not in nature and it is not preferable for listening, crosstalk cancellation processing is not performed when listening to binaural sound signals even with headphones.

[0108] Therefore, as shown in FIG. 3, the audio signal outputted from the audio decoding section 26 is outputted to the audio output terminal 37 c without going through the crosstalk canceler 27. The audio signal output from the audio output terminal 37 c is supplied to the headphone 55. As described above, the viewer 59 can listen to the crosstalk processed signal outputted from the crosstalk canceler 27 by the speech strength 53, 54, or the

audience 59 can listen to the sound signal not processed by the crosstalk canceler 27 as the headphone 55 You can also listen to.

[0109] A reproduction procedure of the binaural audio signal at the headphone 55 will be described with reference to FIG. 27. The same processes as those in the flowchart shown in FIG. 13 are denoted by the same reference numerals, and description thereof is omitted.

[0110] In FIG. 27, steps S181 to S186 are as described with reference to FIG. 13, and in step S192 it is determined whether or not reproduction is performed by the headphone 55. FIG. It is sufficient to detect by a connection detecting unit whether or not a plug is inserted into the sound output terminal 37c which is a connection terminal of the headphone 55. FIG. If it is determined in step S192 that the headphone is to be reproduced (YES), in step S193, a binaural audio signal which is not subjected to crosstalk cancellation processing in synchronization with the video signal is reproduced by the headphone 55, and in step S190 Transfer. Similarly, when the reproduced audio signal is not binaural audio signal but normal stereo sound, the audio signal outputted from the audio decoding section 26 may be similarly supplied to the headphone 55.

[0111] If it is not determined in step S192 that the headphone is to be reproduced (NO), the process of steps S187 to S190 is performed as in FIG. Unlike the reproduction process by the headphone 55 in step S193, the process in step S189 is a process of reproducing the binaural audio signal by the speakers 53,

[0112] ⟨⟨ Second Embodiment ⟩⟩

The photographer 300 attaches the binaural microphone 3 to the left and right ears 302 and picks up the subject and records on the recording medium 44 while collecting the sound, so that the viewer 59 can recognize the entirety when the photographer 300 shoots You can listen to the ambient sound in the direction. However, an image shot by a normal video / audio recording / reproducing apparatus (video camera) is an image in the range of about 60 degrees in front of the camera. In the case of zoom photographing, photographing is performed with a narrower viewing angle. Therefore, in the case of performing zoom shooting, it is desirable to enhance the sound produced in the vicinity of the zoomed up subject and collect the sound. In the second embodiment, in addition to the zoom-up of the subject, the sound in the vicinity of the subject is enhanced, and collected and recorded.

[0113] FIG. 28 shows the video / audio recording / reproducing apparatus 102 provided with the audio processing unit of audio which is the second embodiment, and its configuration and operation will be described. Components having the same functions as those of the video / audio recording / reproducing apparatus 101 of the first embodiment shown in FIG. 3 are denoted by the same reference numerals, and description thereof is omitted. The video / audio recording / reproducing apparatus 102 is different from the video / audio recording / reproducing apparatus 101 in that it includes an audio zoom processing section 33. In FIG. 28, illustration of the audio output terminal 37 c which is the connection terminal of the headphone 55 and the headphone 55 is omitted.

[0114] In FIG. 28, the audio signal from the binaural microphone 3 input via the external microphone connection terminal 32 is input to the audio zoom processing unit 33. The camera unit 11 includes a plurality of lenses (not shown), and has a zooming function of zooming up and down the subject by moving one or a plurality of lenses and varying the distance between the lenses. When a zoom-up operation is performed by the operation unit 48, a zoom-up control signal is input from the control unit 47 to the camera unit 11, the subject is zoomed in and imaged, and the zoom-up control signal is subjected to audio zoom processing The audio signal inputted and input to the unit 33 is subjected to audio zoom processing.

[0115] On the basis of the zoom-up control signal, the audio zoom processing unit 33 augments the audio signal collected on the median plane of the photographer 300 including a lot of audio signals to be emitted from the binaural sound power of the subject, and performs zoom-in Thereby generating an audio signal. The zoom-up audio signal is input to the audio coding section 22 via the switch Swl. The image signal of the subject picked up by zooming up is encoded by the video encoding section 12 and the zoom up audio signal is encoded by the audio encoding section 22 and is recorded on the recording medium 44 as in the first embodiment It is recorded. FIG. 29 is a specific configuration example of the audio zoom processing unit 33. As shown in FIG. 29, the audio zoom processing unit 33 includes a zoom magnification detector 331, a coefficient calculator 332, an adder 335, a variable amplifier 337, and adders 338 a and 338 b.

[0117] In FIG. 29, the zoom magnification detector 331 detects the zoom magnification based on the zoom-up control signal supplied from the control unit 47. The coefficient calculator 332 calculates a coefficient a representing the degree of enhancement of the sound emitted from the vicinity of the subject according to the detected zoom magnification. The adder 355 adds the binaural sound signals of the right and left channels input from the external microphone connection terminal 32. The variable amplifier 337 amplifies the output signal of the adder 335 according to the coefficient α from the coefficient calculator 332. The adders 338 a and 338 b add the left and right channel binaural sound signals to the output signal of the variable amplifier 337, respectively. The diaphragms in the microphones 31a and 31b are substantially parallel in a state in which the microphones 31a and 31b of the binaural microphone 3 are attached to the left and right ears 302 of the photographer 300. Since the sound from the left and right direction of the photographer 300 sometimes includes components having phases opposite to each other, as a result of the addition of the left and right channels in the adder 335, the sound from the left and right directions is partially canceled out and attenuated. Accordingly, from the audio zoom processing unit 33, a zoom-up audio signal in which the sound collected on the median plane of the photographer 300 is enhanced is output.

[0118] With reference to FIG. 30, the operation of the video / audio recording / reproducing apparatus 102 including the operation in the audio zoom processing unit 33 will be further described. 30, in step S201, the adder 335 adds binaural sound signals of left and right channels from the binaural microphone 3 to obtain an addition signal S. In step S undefined 202, the zoom magnification detector 331 detects the zoom magnification obtained by the control unit 47 according to the operation of the operation unit 48. The zoom magnification may be obtained from the relationship between the applied voltage to the motor for driving the lens of the camera unit 1 1 and the driving time. In step S203, the coefficient calculator 332 calculates a coefficient oc representing the enhancement degree according to the zoom magnification. In step S 204, the variable amplifier 337 multiplies the output signal of the adder 335 by the coefficient α to obtain a S. In step S 205, the calorie calculators 338 a and 338 b add the binaural sound signals of the left and right chambers and the output signal (S) of the variable amplifier 337. Then, in step S 206, the zoom-up audio signal is recorded on the recording medium 44. In step S207, the control unit 47 determines whether the recording has ended. If it has not ended (NO), the control unit 47 repeats the force of step S201. If recording has ended in step S207 (YES), the processing in the audio zoom processing unit 33 is ended.

[0119] FIG. 31 shows an audio zoom processing unit 33 a showing another configuration example of the audio zoom processing unit 33. If a head related transfer function that gives the effect that the sound source approaches the listener with respect to the sound signal of the median plane is convoluted, the sound source also receives a power in conjunction with the zoom-up of the subject in the camera unit 11 It is felt as if it approached the listener, and it is possible to receive and listen to the audio signal having the realistic feeling. The audio zoom processing unit 33 a shown in FIG. 31 is configured to convolve a head related transfer function that provides an effect of bringing the sound source closer to the sound signal of the median plane.

[0120] The audio zoom processing unit 33 a shown in FIG. 31 differs from the audio zoom processing unit 33 of FIG. 29 in that a function acquiring unit 333, a transfer function memory 334, and a convolution operator 336 are added.

[0121] In FIG. 31, a transfer function memory 334 stores a head related transfer function for forming a virtual sound source in which a sound source is virtually positioned in close proximity. The head related transfer function is a function to obtain the listening characteristics of the sound emitted from the virtual sound source and the listening characteristics are determined by the distance between the virtual sound source and the listener.

[0122] The function acquiring unit 333 acquires from the transfer function memory 334 a head related transfer function corresponding to a sound localization position assumed according to the coefficient oc calculated by the coefficient calculator 332. Note that the coefficient α in FIG. 29 and the coefficient α in FIG. 31 and the other figures are not necessarily the same. For the sake of convenience, the same reference numerals are attached. The convolution arithmetic unit 336 convolves the head related transfer function acquired by the function acquiring unit 333 with the addition signal of the binaural sound generated by the adder 335. The variable amplifier 337 amplifies the added signal convoluted with the head-related transfer function according to the coefficient α obtained by the coefficient calculator 33 2. Calo calculators 338 a and 338 b add the left and right channel's neural audio signals and the variable amplifier 337 output signals. Here, it is sufficiently effective only to locate the force sound source having the variable amplifier 337 virtually in close proximity, and it is also possible to omit the variable amplifier 337. The coefficient α for selecting the head-related transfer function by the function acquiring unit 333 and the coefficient determining the degree of amplification in the

variable amplifier 337 may be set as different coefficients. [0123] With reference to FIG. 32, A method of measuring the head related transfer function for forming a virtual sound source will be explained.

[0124] The head-related transfer function measuring apparatus 6 a shown in FIG. 32 is configured to include a personal computer 61, an amplifier 62, a speaker 63, amplifiers 66 a and 66 b, and a dummy head microphone **68. Dummy one head microphone 68 í, artificial head 681** 【Install the microphone units 684 a, 684 b! The cranial transfer function measuring device 6a uses a dummy head microphone 68 instead of the microphone units 65a and 65b attached to the cylindrical structure 65e, as compared with the head related transfer function measuring device 6 shown in FIG. 15 Except that the point and the speaker are arranged on the median plane of the dummy head microphone 68 as only one of the left and right speakers 63 and 64 (in this case, the speaker 63 is used)

[0125] FIG. 33 shows a sectional view of the dummy head microphone 68. In the dummy head microphone 68, micro switches 684 a and 684 b are disposed in the vicinity of entrances of the auditory canals 683 a and 683 b of the artificial head 681 having the pinna shaped members 682 a and 682 b. The dummy head microphone 69 shown in FIG. 16 (C) is obtained by disposing the microphone units 694 a, 694 b at positions corresponding to the position of the eardrum of the person in the back of the ear canals 693 a, 693 b, and the dummy head microphone 68 , The dummy head microphone 69 is different from the dummy head microphone 69 in that the micro units 684 a and 684 b are arranged near the entrances of the auditory canals 683 a and 683 b. Normally, the dummy head microphone refers to one in which the microphone units 694 a and 694 b are arranged at positions corresponding to the position of the eardrum of the person at the back of the ear canals 693 a and 693 b as shown in FIG. 16 (C) 33 in which the microphone units 684a, 684b are disposed in the vicinity of the population of the auditory canals 683a, 683b of the artificial head 681 having the 1S auricular shapes 682a, 682b is also referred to as a dummy head microphone for convenience.

[0126] By using the dummy head microphone 68, it is possible to collect sound from the speaker 63 as binaural sound excluding the influence by the auditory canal 683 a, 683 b.

[0127] Returning to FIG. 32, the personal computer 61 generates a measurement signal composed of impulse sound, for example. The measurement signal is amplified by the amplifier 62. Measurement signals generated from the speaker 63 are received by the left and right microphone units 684 a, 684 b of the dummy head microphone 68. The left and right signals obtained by receiving the sound are amplified by the amplifiers 66a and 66b and input to the personal computer 6 1. The personal computer 61 compares the generated measurement signal with the received signal to determine the head related transfer functions h (t), h (t) of the dummy head microphone 68. Head

1 r

The part transfer function h (t) is the head related transfer obtained by the sound receiving signal at the left microphone unit 684a

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The function, head related transfer function h (t) is a head related transfer function obtained from a sound receiving signal at the right microphone unit 684 b. The distance D between the speaker 63 and the dummy head microphone 68 is set to, for example, 0.5 m, 1 m, 2 m, etc., and head related transfer functions at each distance D are sequentially found.

[0128] 34 to 39 show characteristics of the head related transfer function obtained by measuring with the head-related transfer function measuring device 6 a shown in FIG. 32.

[0129] The impulse response waveform shown in FIG. 34 (A) is a waveform received by the left microphone unit 684 a when the distance D between the speaker 63 and the dummy head microphone 68 is 50 cm. The vertical axis is the normalized amplitude (voltage). The horizontal axis is time, and it is indicated by the number of sample points of the signal whose sampling frequency is 48 kHz. FIG. 34 (B) is a frequency response characteristic obtained by Fourier analysis of the impulse response waveform shown in FIG. 34 (A) by the personal computer 61. The horizontal axis is the frequency (Hz), and the vertical axis is the response characteristic.

[0130] FIG. 35 (A) shows the impulse response waveform received by the right microphone unit 684 b when the distance D is 50 cm. FIG. 35 (B) is a frequency response characteristic obtained by Fourier analysis of the impulse response waveform shown in FIG. 35 (A). The measurement conditions are the same as in FIG. 34.

[0131] Similarly, FIG. 36 (A) shows an impulse response waveform received by the left microphone unit 684 a when the distance D is 1 m, and FIG. 36 (B) shows its frequency response characteristic.

[0132] FIG. 37 (A) shows the impulse response waveform received by the right microphone unit 684 b when the distance D is 1 m, and FIG. 37 (B) its frequency response characteristic.

[0133] FIG. 38 (A) shows the impulse response waveform received by the left microphone unit 684 a when the distance D is 2 m, and FIG. 38 (B) its frequency response characteristic.

[0134] FIG. 39 (A) shows the impulse response waveform received by the right microphone unit 684 b when the distance D is 2 m, and FIG. 39 (B) its frequency response characteristic.

[0135] Comparing these characteristics, the amplitude of the waveform decreases as the impulse response waveform shown in (A) of FIGS. 34 to 39 increases the value of the distance D to 0.5 m, 1 m, 2 m. In the frequency response characteristic shown in (B) of FIG. 34 to FIG. 39, when the distance D is 0.5 m, regular concavo-convex characteristics at intervals of about 400 Hz in the frequency portion of lkHz to 4 kHz enclosed by the ellipse of a broken line Dip), and when the distance D is lm, it becomes somewhat irregular asperity characteristics, and when the distance D is 2 m, it is a combination of a plurality of concavo-convex characteristics with different frequency intervals. In addition, when the distances D are the same, substantially the same characteristics are obtained on the left and right.

[0136] The personal computer 61 compares the measurement signal of the generated impulse sound with the waveform of the impulse response signal input from the amplifiers 66a and 66b to obtain head related transfer characteristics for each distance D. The head related transfer characteristic obtained for each distance D is a characteristic for causing the listener to listen to the sound signal of the virtual sound source force by locating the sound source virtually at the position of the distance D. In this embodiment, the distances D are set to 0.5 m, 1 m, and 2 m, but even if the respective distances D may be determined to be narrower than 0.5 m even if the respective characteristics are obtained at more distances Good.

[0137] The head related transfer characteristics obtained as described above are stored in the transfer function memory 334 of FIG. Which of the stored transfer functions is to be used for the zoom magnification detected by the zoom magnification detector 331 is determined as an example by setting the distance to the subject measured **by the automatic focus measuring function Use the coefficient α obtained by dividing by the zoom** magnification. For example, when the distance to the subject is 10 m and the zoom magnification is 5, the **coefficient is 2. When the distance to the subject is 10 m and the zoom magnification is 10, the coefficient α is 1, and when the zoom magnification is 20, the coefficient α is 0.5.**

[0138] With reference to FIG. 40, the operation of the video / audio recording / reproducing apparatus 102 including the operation at the audio zoom processing unit 33 a will be further described. 40, in step S211, the calorifier 335 adds binaural sound signals of left and right channels from the binaural microphone 3 to obtain a calorie calculation signal S. In step S undefined 212, the zoom magnification detector 331 detects the zoom magnification obtained by the control unit 47 according to the operation of the operation unit 48. In step S213, the coefficient calculator 332 determines which one of the plurality of transfer functions stored in the transfer **function memory 334 is selected according to the zoom magnification, and determines the coefficient α** representing the degree of enhancement in the variable amplifier 337 . As the coefficient OC, the number obtained by dividing the distance to the subject by the zoom magnification may be used as it is, or the value generated based on the number obtained by dividing the distance to the subject by the zoom magnification Even if it is! /,.

[0139] In step S 214, the function acquiring unit 333 acquires a transfer function from the transfer function memory 334 according to the coefficient oc, and the convolution operator 336 convolutes the transfer function with the addition signal output from the adder 335. In step S215, the variable amplifier 337 multiplies the output **signal of the convolution arithmetic unit 336 by the coefficient α to enhance it. In step S 216, the adders 338 a** and 338 b add the binaural sound signals of the left and right channels and the output signal of the variable amplifier 337. Then, in step S 216, the zoom-up audio signal is recorded on the recording medium 44. In step S218, the control unit 47 determines the force at which recording has ended, and if not completed (NO), repeats

the force of step S211. If the recording has ended (YES) in step S218, the processing in the audio zoom processing section 33a is ended.

[0140] ⟨⟨ Third embodiment ⟩⟩

In the second embodiment, the zoom-up process of audio is performed on the recording side, but in the third embodiment, the zoom-up process of audio is performed on the reproduction side. In the video / audio recording / reproducing apparatus 103 according to the third embodiment shown in FIG. 41, the same reference numerals are given to components having the same functions as the video / audio recording / reproducing apparatus 101 of the first embodiment shown in FIG. 3 And its explanation will be omitted. Compared with the video / audio recording / reproducing apparatus 101, the video / audio recording / reproducing apparatus 103 has the audio zoom processing section 33 b at the subsequent stage of the separation section 15, and the zoom magnification detector 331 is provided at the front stage of the multiplexing section 13 The difference is that it is. In FIG. 41, illustration of the audio output terminal 37 c which is a connection terminal of the headphone 55 and the headphone 55 is omitted.

[0141] The operation of the video / audio recording / reproducing apparatus 103 will be described. The operation unit 48 is operated, and the lens drive signal of the camera unit 11 generated by the control unit 47 is supplied to the camera unit 11 and the zoom magnification detector 331. The zoom magnification detector 331 detects the zoom magnification by analyzing the zoom direction, the zoom speed, and the lens driving time of the lens driving signal. The zoom magnification information indicating the detected zoom magnification is supplied to the multiplexing unit 13. The multiplexing unit 13 also multiplexes the zoom magnification information together with the encoded video signal, the encoded audio signal, and the binaural flag signal. The recording / reproducing unit 14 records the multiplexed signal including the zoom magnification information on the recording medium 44.

[0142] The recording / reproducing unit 14 reproduces the multiplexed signal recorded on the recording medium 44, and the demultiplexing unit 15 separates the zoom magnification information together with the coded video signal, the encoded voice signal and the binaural flag signal . The zoom magnification information is input to the audio zoom processing unit 33 b.

[0143] FIG. 42 is a specific configuration example of the audio zoom processing unit 33 b. As shown in FIG. 42, the audio zoom processing unit 33 b differs from the audio zoom processing unit 33 of FIG. 29 in that the zoom magnification detector 33 1 is eliminated and the signals input to the adders 338 a and 338 b are But is different from the output signal of the crosstalk canceller 27.

[0144] In FIG. 42, the coefficient calculator 332 calculates a coefficient α used when the input signal is amplified by the variable amplifier 337 using the zoom magnification information separated and input by the demultiplexer 15. The adder 335 adds the noisy speech signal input from the audio decoding unit 26. The variable amplifier 337 amplifies the output signal from the adder 335 according to the coefficient α input from the coefficient calculator 332. The adders 338 a and 338 b add the output signal from the crosstalk canceller 27 and the amplified signal from the variable amplifier 337.

[0145] Meanwhile, there are cases where zoom processing in the camera unit 11 is executed by computation processing by software using, for example, DSP. When the audio zoom processing is performed at the time of reproduction, it is unnecessary to secure the signal processing time of the DSP necessary for the zoom processing. Therefore, it is possible to sufficiently perform signal processing such as optimization processing of a video signal photographed by a DSP at the time of recording, encoding of a video signal, recording control or the like. By performing the audio zoom processing at the time of reproduction, it is possible to avoid the possibility of occurrence of a problem that the calculation man-hours at the DSP are assigned to the calculation processing for the zoom processing and the operation time for recording is insufficient Show.

[0146] 43 shows an audio zoom processing unit 33 c configured to convolve a head related transfer function that gives the proximity effect to the audio signal of the median plane to the audio zoom processing unit 33 b of FIG. 42 ing. The audio zoom processing unit 33 c differs from the audio zoom processing unit 33 b in that a function acquiring unit 333, a transfer function memory 334, and a convolution operator 336 are added. The functions of the function acquiring unit 333, the transfer function memory 334, and the convolution computing unit 336 are the same as those in FIG. 31, and description thereof is omitted.

[0147] ⟨⟨ Fourth Embodiment ⟩⟩

The video / audio recording / reproducing apparatus 104 according to the fourth embodiment shown in FIG. 44 is configured so that the audio zoom processing is performed manually by an external operation at the time of reproducing the recording medium 44. That is, when the zoom magnification information is not recorded on the recording medium 44, the viewer 59 performs the zoom up processing of the audio while watching the video signal reproduced by the motor 52. The zoom-up process of the audio manually performed by the viewer 59 is referred to as manual audio zoom process.

[0148] Compared with the video / audio recording / reproducing apparatus 103, the video / audio recording / reproducing apparatus 104 shown in FIG. 44 omits the zoom magnification detector 331, has the audio zoom processing section 33 d instead of the audio zoom processing section 33 b, Different in that! /.

[0149] When the viewer 59 instructs the operation unit 48 to perform manual audio zooming, the control unit 47 inputs the zoom-up control signal to the audio zoom processing unit 33 d. The audio zoom processing unit 33 d performs zoom-up processing on the binaural audio signal decoded by the audio decoding unit 26 according to the zoom-up control signal.

[0150] FIG. 45 is a specific configuration example of the audio zoom processing unit 33 d. As shown in FIG. 45, the audio zoom processing unit 33 d is provided with a zoom magnification detector 331 a to which the zoom-up control signal from the control unit 47 is input, in comparison with the zoom processing unit 33 b in FIG. 42, and coefficient calculation But differs in that it inputs the zoom magnification information generated by the zoom magnification detector 33 la instead of the zoom magnification information from the separation section 15 to the unit 332. The other parts are the same operations as those of the zoom processing section 33 b, and the description thereof will be omitted.

[0151] The manual audio zoom process in the fourth embodiment will be further described with reference to FIG. 46. In FIG. 46, in step S221, the adder 335 adds the right and left signals of the binalolar audio signal obtained by reproduction and obtains the addition signal S. In step S undefined 222, the control unit 47 determines whether or not the audio zoom magnification is varied by the operation unit 48. If the audio zoom magnification is varied in step S222 (YES), the process goes to step S223, and if not changed (NO), the process goes to step S226.

[0152] When the audio zoom magnification is varied, the zoom magnification detector 33 la calculates the zoom magnification based on the zoom-up control signal in step S223. In step S224, the coefficient calculator 332 calculates a coefficient (X) based on the zoom magnification input from the zoom magnification detector 331a, and the coefficient (X is a head for locating the sound source in front of the viewer But may also include the transfer function characteristic In step S225, the value of the coefficient α is updated to the newly calculated value.

[0153] In step S226, the variable amplifier 337 multiplies the added signal S by the coefficient α to generate a S. In the case where the steps S223 to S225 have not passed, the coefficient α is a value before the audio zoom magnification is varied. In step S227, the adders 338a and 338b add the signal "S" to the binaural sound signal subjected to the crosstalk cancellation processing by the crosstalk canceller 27. Then, In step S228, the audio signal obtained in step S227 is output via the switch Sw2 and the audio output terminal 37b. In step S229, the control unit 47 determines whether or not the reproduction has ended. If it is not completed (NO), the control unit 47 returns to step S22 1, and if it is completed (YES), ends the processing.

[0154] ⟨⟨ Fifth Embodiment ⟩⟩

The video / audio recording / reproducing apparatus 105 of the fifth embodiment shown in FIG. 47 shows a configuration suitable for listening to the zoom-up audio signal with the headphone 55. The video / audio recording / reproducing apparatus 105 shown in FIG. 47 is different from the video / audio recording / reproducing apparatus 103 shown in FIG. 41 in that it has an audio zoom processing section 33 e instead of the audio zoom processing section 33 b, and the audio / 33 e are supplied to the headphone 55 via the audio output terminal 37 c. The crosstalk cancellation processing is not performed on the headphone 55 by the crosstalk canceller 27, and the binaural audio signal subjected to the zoom-up processing by the audio zoom processing unit 33 e is supplied.

[0155] FIG. 48 is a specific configuration example of the audio zoom processing unit 33 e. The audio zoom processing unit 33 e differs from the audio zoom processing unit 33 b of FIG. 42 in that adders 338 c and 338 d

are added. The adders 338 c and 338 d add the zoom-up sound signal output from the variable amplifier 337 to the binaural sound signal obtained by decoding by the audio decoding section 26. The headphone listening audio signal added by the adders 338 c and 338 d is supplied to the headphone 55 via the audio output terminal 37 c.

[0156] The effect of reproducing the zoom-up audio signal according to the second to fifth embodiments described above is as follows.

[0157] When the zoom magnification is set to be small, the signal added by the adder 335 is not increased by the variable amplifier 337, so that the viewer 59 sets the image signal displayed on the monitor 52 and the photographing And the audio signal having realistic sensation reproduced by 360 degrees audio signal surrounding the person 300 with the speakers 53 and 54. Even when the viewing angle is set wide, the viewing angle is about 60 degrees. The viewer 59 sometimes recognizes that it is a so-called blank speech sound in which the sound generated by the subject force displayed on the monitor 52 is insufficient due to the difference between the viewing angle of the image and the angle range when collecting the sound signal . On the other hand, as described above, since the audio signal subjected to the audio zoom-up process is enhanced for the photographer 300 by adding the signal component from the median plane to the binaural sound signal, the blank area is compensated Sound signal. Therefore, it is possible to prevent the viewer 59 from recognizing that the viewer 59 is a hollow speech sound, and the viewer 59 can listen to a sound having a realistic feeling with less sense of discomfort as compared with the first embodiment.

[0158] ⟨⟨ Sixth Embodiment ⟩⟩

The video / audio recording / reproducing apparatus 106 according to the sixth embodiment shown in FIGS. 49 and 50 is not provided with built-in stereo microphone 21 and binaural microphone 3 separately as in the first to fifth embodiments Shows a configuration in which a normal stereophonic sound collecting microphone and a binaural microphone are shared. 49 is a plan view showing an external configuration example of the video / audio recording / reproducing apparatus 106 in the sixth embodiment, and FIG. 50 is a block diagram showing a concrete internal configuration example of the video / audio recording / reproducing apparatus 106. In FIGS. 49 and 50, components having the same functions as those in FIGS. 1 and 3 are denoted by the same reference numerals, and description thereof is omitted.

[0159] As shown in FIG. 49, the video / audio recording / reproducing apparatus 106 houses the microphone placing sections 35 a, 35 b for placing the microphones 31 e, 31 f on the top surface and the microphone cords 310 e, 310 f connected to the microphones 31 e, 31 f And a cord storage section 34.

[0160] In FIG. 49, when sound is collected by normal stereo sound using the microphones 31 e, 31 f, the photographer 300 places the microphones 3 le, 3 If on the microphone placing parts 35 a, 35 b and collects sound. In the case of collecting binaural sounds, the photographer 300 pulls out the microphone cords 310e and 310f from the cord storage unit 34, attaches the microphones 31e and 31f to the ear 302, and collects sounds. The video / audio recording / reproducing apparatus 106 has a projection-like detecting section (not shown) for detecting the microphones 31 e, 31 f placed on the microphone placing sections 35 a, 35 b. With the on-Z off of the switch (corresponding to the switch Sw 4 in FIG. 50) that operates in conjunction with the protrusion detector, the video / audio recording / reproducing device 106 places the microphones 3 le, 31 f on the microphone placing portions 35 a, 35 b Whether or not the force is being detected. The detection of whether or not the force placed on the microphone placement portions 35a and 35b is detected is not limited to this, and for example, the magnetic field generating also the permanent magnet force built in the microphones 31e and 31f may be detected by a Hall element or a magneto- It is also possible to detect using.

[0161] Specifically, in FIG. 50, if the microphones 31 e and 31 f are not placed on the microphone placing parts 35 a and 35 b, the switch Sw 4 is connected to the terminal e and turned off, the microphones 31 e and 31 f are placed If it is connected to the terminal f, it is turned on. The placement detection section 41a detects whether or not the microphone 31e, 31f is placed on the microphone placing sections 35a, 35b, depending on whether the switch Sw 4 is connected to the force terminal f connected to the terminal e . The detection signal from the placement detection section 41 a is supplied to the control section 47.

[0162] When the placement detection unit 41a detects that the microphones 31e and 31f are placed, the microphones 31e and 3 If perform normal stereo sound collection, and the control unit 47 controls the video / audio recording / reproducing apparatus 106 controls each part of the circuit so as to perform a recording operation corresponding to photographing in the normal mode. In this case, the roles of the microphones 31 e and 31 f are equivalent to the built-in stereo microphone 21 in FIG. 3, and the flag information generating unit

42 does not generate the binaural flag signal indicating that it is in the binaural mode. On the other hand, when the placing / detecting unit 41 a detects that the microphones 31 e and 31 f are not placed, the photographer 300 wears the microphones 31 e and 3 If on the ears 302 to collect binaural sounds in binaural mode , The control unit 47 controls each unit of the circuit so that the video / audio recording / reproducing apparatus 106 performs the recording operation corresponding to the imaging in the binaural mode. In this case, the flag information generating unit 42 generates a no-noal flag signal under the control of the control unit 47.

[0163] In the sixth embodiment, the microphones 31 e and 31 f, the microphone placing parts 35 a and 35 b, the placing and detecting part 41 a, and the control part 47 as a whole, as microphones for collecting ambient sounds, a bike to be worn by the photographer's ears It acts as a switching unit that switches whether to use a microphone other than a binaural microphone that uses a nor- mal microphone.

[0164] Here, an example of the structure of the cord storage section 34 will be described with reference to FIG. 51. 51 (A) shows the internal structure of the cord storage portion 34 with respect to the upper surface side force, and the microphone cords 310 e and 310 f are wound around the reel 341 having the rotation shaft 343. 51 (B) shows an internal structure of the cord storage portion 34 as viewed from the lower surface side, and a spiral spring 342 is provided inside the reel 341. Instead of or in addition to detecting whether or not the microphones 31e and 31f are placed on the microphone placing parts 35a and 35b, by detecting the rotation angle of the reel 341, it is possible to determine whether or not the binaural mode You can also detect.

[0165] ⟨⟨ Seventh embodiment ⟩⟩

The video / audio recording / reproducing apparatus 107 of the seventh embodiment shown in FIG. 52 shows a configuration using a wireless type binaural microphone that wirelessly transmits the sound signal obtained by collecting the sound signal to the main body of the apparatus. In FIG. 52, components having the same functions as those in FIG. 1 are denoted by the same reference numerals, and description thereof is omitted.

[0166] In FIG. 52, the video / audio recording / reproducing apparatus 107 includes a radio transmission / reception section 39 instead of the external microphone connection terminal 32 of FIG. 1, and performs voice recording using a radio type binaural microphone 38 instead of the noise noisy microphone 3 . The photographer 300 attaches the wireless binaural microphone 38 wirelessly coupled to the main body of the apparatus to the head and inserts the right and left microphones 38 a and 38 b into the ear 302 to collect the sound so that it is not interrupted by the microphone code You can shoot. It is also possible for two persons, the photographer 300 and a sound collector (not shown) to take photographs.

[0167] With reference to FIG. 53, the internal configuration of the video / audio recording / reproducing apparatus 107 will be described. In FIG. 53, components having the same functions as those in FIG. 3 are denoted by the same reference numerals, and description thereof is omitted. 53 is different from the video / audio recording / reproducing apparatus 10 1 shown in FIG. 3 in that a wireless transmitting / receiving unit 39 is provided instead of the external microphone connecting terminal 32 and the connection detecting unit 41 Different in point!

[0168] When the wireless transmitting / receiving unit 39 receives the binaural sound signal from the wireless type binaural microphone 38, the control unit 47 determines that the switch Swl is connected to the terminal b And causes the binaural audio signal to be supplied from the wireless type binaural microphone 38 to the audio encoding section 22. In this case, the control unit 47 controls the flag information generating unit 42 to generate a binaural flag signal. Further, if it is determined that the wireless main binaural microphone 38 also exceeds the predetermined distance, the control unit 47 connects the switch Swl to the terminal a and transmits the stereo sound signal from the built-in stereo microphone 21 to the terminal a And supplies it to the audio encoding section 22. In this case, the flag information generating unit 42 does not generate the binaural flag signal.

[0169] FIG. 54 shows an example of the internal configuration of the wireless binaural microphone 38 and the wireless transceiver 39, and their operation will be described.

[0170] As shown in FIG. 54, the microphone 38 a of the wireless type binaural microphone 38 includes a microphone unit 38 1, a microphone amplifier 382, a transmission / reception unit 383, an antenna 384, and a warning signal transmission unit 385. The force microphone 38 b (not shown) has the same configuration as that of the microphone 38 a except that it does not include the warning signal transmitting unit 385. The wireless transceiver 39 includes a transmission / reception unit 391, a microphone determination unit 392, a distance

measurement unit 393, a communication coverage area determination unit 394, a warning signal transmission unit 395, and an antenna 396.

[0171] The microphone unit 381 of the microphone 38 a (38 b) generates a binaural sound signal. The microphone amplifier 382 amplifies the noisy audio signal from the microphone unit 381. The transmission / reception unit 383 modulates the amplified binaural sound signal from the microphone amplifier 382 according to a predetermined modulation scheme, and transmits the modulated binaural sound signal via the antenna 384. The warning signal transmitting unit 385 generates an alarm signal based on the warning signal generated by the warning signal transmitting unit 395 of the wireless transmitting / receiving unit 39 (described later) and transmitted via the transmitting / receiving unit 391 and the transmitting / receiving unit 383.

[0172] The antenna 396 of the wireless transceiver 39 receives the modulated signals transmitted from the right and left microphones 38 a and 38 b. The transmitting / receiving unit 391 demodulates the received modulated signal to generate a binaural sound signal and measures the intensity of the received power of the modulated signal. The distance measuring unit 393 estimates the distance from the wireless transmitting / receiving unit 39 to the wireless binaural microphone 38 based on the measured intensity of the received power. The communicable range inside determination unit 394 determines whether the estimated distance is a force within a communicable predetermined distance. The determination result of the communication coverage area determination unit 394 is supplied to the control unit 47. If the estimated distance is within a predetermined communication distance, the control unit 47 connects the switch Swl to the terminal b, and controls the flag information generation unit 42 to generate a binaural flag signal. The control unit 47 connects the switch Swl to the terminal "a" when the estimated distance exceeds a predetermined distance capable of communicating.

[0173] The warning signal transmitting unit 395 generates a warning signal when it is determined that the distance estimated by the communication range in-coverage determining unit 394 exceeds a predetermined distance that can be communicated. The warning signal is supplied to the control unit 47. The control unit 47 generates a warning mark and supplies it to the display unit 17 to display a predetermined warning mark on the display unit 17. When the warning signal transmitting unit 395 does not generate the warning signal, the microphone judging unit 392 judges that the binaural sound signal is normally obtained, and the binaural sound signal obtained by demodulating by the transmitting / And supplies it to the audio encoder 22 via the switch Swl.

[0174] FIG. 55 shows an example of a warning display performed by the wireless type binaural microphone 38 and the video / audio recording / reproducing apparatus 107. The microphone 38 a is provided with a rod-like member having a light emitting diode (LED) 386 attached to its distal end portion. A warning signal generated by the warning signal transmitting unit 385 is supplied to the LED 386, and the LED 386 blinks (or lights) based on the warning signal. In addition to blinking LED 386, or LED 386 may flash warning sound instead of blinking! In this case, it is possible to lower the level of the warning sound or to set the frequency of the warning sound to a low frequency of several tens of Hz or less, for example, so that the warning sound is not heard by the microphone unit 381 Preferable,.

[0175] The warning signal transmitter 395 generates a warning signal when it is determined that the wireless binaural microphone 38 has crossed the communicable area indicated by the dashed circle. As shown in FIG. 55, if the wireless type binaural microphone 38 exceeds the communicable area, a predetermined warning mark is displayed on the display unit 17.

[0176] FIG. 56 shows an example of a warning mark displayed on the display unit 17. The warning mark 171 a shown in FIG. 56 (A) is displayed with the X mark superimposed on the no-noal mark 171 shown in FIG. 5 (A). The warning mark 172a shown in FIG. 56 (B) is obtained by graying out the mark 172 shown in FIG. 5 (B). Any of the marks in FIGS. 56 (A) and 56 (B) may be used as the warning mark, and other marks may be used as a matter of course. If it is assumed that the received power at the wireless transmission / reception unit 39 is below the reception limit despite the warning display, the control unit 47 switches from the wireless binaural microphone 38 to the built-in stereo microphone 21.

[0177] The operation of the video / audio recording / reproducing apparatus 107 will be described further using FIG. 57. In FIG. 57, in step S251, the control unit 47 determines whether it is in binaural mode. If it is not determined in step S251 that the binaural mode is set (NO), in step S253, the recording and reproducing unit 14 collects the sound by the built-in stereo microphone 21 and records a normal stereo sound signal on the recording medium 44. FIG. If the binaural mode is determined in step S251 (YES), in step S252, the wireless transceiver 39 receives the transmission signal from the wireless binaural microphone 38. In step S undefined

254, the distance measuring unit 393 detects the distance from the wireless transmitting / receiving unit 39 to the wireless binaural microphone 38 based on the strength (receiving intensity) of the received power. In step S255, the communication coverage area determination unit 394 determines whether or not the detected distance is within a predetermined distance.

[0178] If it is not determined in step S255 that the distance is within the predetermined distance (NO), the control unit 47 determines in step S257 whether or not the warning display time t is 0 (non-present) and determines whether or not the warning display time If t is 0, the warning signal transmitting unit 395 is controlled to generate a warning signal in step S300. After the warning signal is generated, the process returns to step S254 to repeat the same process. In step S 257, if the warning display time t is not 0 (NO), in step S 258, the control unit 47 determines whether the warning display time t exceeds a predetermined maximum time t max or not and determines the maximum If the time tmax has not been exceeded (NO), the process proceeds to step S300, and the process returns to step S254 in the same manner. If it exceeds the maximum time tmax (YES), in step S259, the control unit 47 controls the switch Swl so as to switch from the wireless type binaural lull microphone 38 to the built-in stereo microphone 21, and stops the generation of the warning signal So as to control the warning signal transmitter 395. Then, the process proceeds to step S25 3.

[0179] If it is determined in step S255 that the distance is within the predetermined distance (YES), in step S256, if the warning signal transmitting unit 395 is generating a warning signal, the control unit 47 stops generating the warning signal As shown in FIG. Then, in step S 301, the recording 'playback unit 14 collects the sound by the wireless type binaural microphone 38 and records the binaural sound signal on the recording medium 44. In step S302, the control unit 47 determines whether or not an operation for ending the recording has been performed. If the operation for ending the recording has not been performed (NO), the control unit 47 returns to step S251, and if the recording end operation has been performed (YES), the process is terminated.

Industrial applicability

[0180] The present invention can be applied not only to consumer video cameras but also to professional video cameras to video and audio recording and reproducing apparatuses that need to reproduce realistic sounds together with the captured images. It can also be applied to digital cameras and mobile phones with movie shooting functions. Furthermore, although the present invention is suitable for use in a video / audio recording / reproducing apparatus for recording and reproducing both a video signal and an audio signal, it is sufficiently effective even when used in an audio recording / reproducing apparatus for recording / reproducing only an audio signal .

WO2006054698A1 VIDEO/AUDIO RECORDING APPARATUS AND METHOD, AND VIDEO/AUDIO REPRODUCING APPARATUS AND METHOD

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The wording below is an initial machine translation of the original publication. To generate a version using the latest translation technology, go to the original language text and use Patent Translate.

The scope of the claims

[1] (101, 102, 103, 104, 105, 106) for recording a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer (300) including a sound emitted from the subject, 107) 【こっ!/ヽて,

A camera unit (11) for imaging the object,

A switch unit (Sw1) for switching whether to use a microphone other than the binaural microphone (3) using the binaural microphone (3) worn on the ear of the photographer (300) as the ambient sound)When,

A video processing section (12) for processing the video signal outputted from the camera section (11), an audio processing section (22) for processing the audio signal outputted from the microphone which sounds the ambient sound,

A sound mode when the ambient sound is turned on as a binaural mode when switching to the binaural microphone (3) as a microphone for collecting the ambient sound by the switching unit (Swl), and that the sound mode is a noorrhard mode A flag information generating section (42) for generating a binaural flag signal indicating a binaural flag

A recording unit (14) for recording the video signal processed by the video processing unit (12), the audio signal processed by the audio processing unit (22), and the binaural flag signal on a recording medium,

(101, 102, 103, 104, 105, 107) characterized in that it comprises:

[2] A built-in microphone (21) incorporated in the video and audio recording apparatus (101, 102, 103, 104, 105, 107)

An external microphone connection terminal (32)

A setting section (48, 47) for setting as the external microphone to be connected to the external microphone connection terminal (32) whether to set the binaural lull mike (3) or a microphone other than the binaural microphone; a setting section A connection detection section (41) for detecting whether or not an external microphone is connected to the connection section

A switch (Swl) for switching between the audio signal output from the built-in microphone (21) and the audio signal output from the external microphone as audio signals to be supplied to the audio processing unit (22)

It is detected that the external microphone is set to the binaural microphone (3) by the setting section (47, 48) and the external microphone is connected to the external microphone connection terminal (32) by the connection detection section (41) , And a control unit (47) for setting the audio mode to the binaural mode,

When the voice mode is the binaural mode, the control unit (47) controls the switch (Sw1) so that the switch (Sw1) supplies the audio signal output from the external microphone to the audio processing unit (22)), And the flag information generating section (42) controls the flag information generating section (42) so as to generate the binaural flag signal as described in the first aspect of the invention Audio recording apparatus (101, 102, 103, 104, 105, 107).

[3] A display unit (17) for displaying a video signal output from the camera unit (11)

Characterized in that it further comprises a display control unit (47) for displaying a binaural mark indicating that the binaural mode is displayed on the display unit (17) when the audio mode is the inno informal mode. The video and audio recording apparatus (101, 102, 103, 104, 105, 107) according to 1.

[4] The camera unit (11) has a zoom function for enlarging and capturing the object,

A sound zoom processing unit (33) for amplifying the sound signal collected by the binaural microphone (3) according to the enlargement ratio in the camera unit (11). (102) according to the first aspect of the invention.

[5] The camera unit (101) has a zoom function for enlarging and capturing the object,

A head related transfer function for forming a virtual sound source by virtually bringing a sound source of an audio signal collected by the binaural microphone (3) close to a listener corresponds to a plurality of distances between the virtual sound source and the listener A transfer function memory (334) for storing and storing a plurality of head related transfer functions stored in the transfer function memory (33 4) selectively according to the enlargement ratio in the camera section (11) A convolution operator (336) for convolving the head related transfer functions acquired by the function acquiring unit (333) with respect to a sound signal collected by the binaural microphone (3), a function acquiring unit (333) (102, 103) according to claim 1, further comprising an audio zoom processing section (33a, 33c) having a plurality of audio signal processing sections (33a, 33b)

[6] A video and audio recording method for recording a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer (300) including a sound emitted from the subject,

An imaging step (S155) of imaging the subject,

A switching step (S 151) of switching whether to use a microphone other than a binaural microphone that uses a binaural microphone (3) worn on the ear of the photographer (300) as the ambient sound,

A video processing step (S156) of processing a video signal obtained by imaging the subject, an audio processing step (S 156) of processing a sound signal output from a microphone for collecting the ambient sound,

Wherein when switching to the binaural rhythmic (3) as a microphone for collecting the ambient sound at the switching step (S 151), a sound mode at the time of collecting the ambient sound is set as a binaural mode, a noiselal mode A flag information generating step (S 158) for generating a binaural flag signal indicating that the binaural flag signal is present,

A recording step (S161) of recording in the recording medium (44) the video signal processed in the video processing step (S156), the audio signal processed in the audio processing step (S156) and the binaural flag signal)When,

And a video / audio recording method.

[7] An audio / video reproducing apparatus (101) for reproducing a recording medium (44) on which a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer (300) including a sound emitted from the subject are recorded , 102, 103, 104, 105, 106, 107) 【Ko!

A reproduction unit (14) for reproducing a recording signal recorded on the recording medium,

A separation unit (15) for separating the video signal and the audio signal from the recording signal reproduced by the reproduction unit (14)

A video processing unit (16) for processing the video signal separated by the separation unit (15), an audio processing unit (26) for processing the audio signal separated by the separation unit (15), ,

When a binaural flag signal indicating that a binaural microphone worn on the ear of the photographer (300) is used as the microphone for collecting the ambient sound on the recording medium (44), the binaural flag A flag information acquisition unit (36) for acquiring a flag signal, a flag information acquisition unit (36) for acquiring a flag signal, a flag information acquisition unit (36) for acquiring a flag signal, And a crosstalk

canceller (27) for processing the audio signal so as to cancel out a crosstalk signal generated when sound is generated by the sound source (53, 54)

The crosstalk canceller (27) has filters (272 a - 272 d) for convolving the filter characteristics obtained in advance with respect to the audio signal, and the filter characteristic is obtained by combining the measurement signal with a pair of microphones (101, 102, 103, 104, 105, 106, 107) characterized in that it is a filter characteristic based on a head related transfer function measured using collected sound signals.

A video and audio reproducing method for reproducing a recording medium (44) in which a video signal obtained by imaging a subject and an audio signal obtained by collecting ambient sound of a photographer (300) including sound emitted from the subject are recorded,

A reproducing step (S 181) of reproducing a recording signal recorded on the recording medium (44); a separating step of separating the video signal and the audio signal from the recording signal reproduced in the reproducing step (S183)

A video processing step (S 184) for processing the video signal separated in the separating step (S 183)

An audio processing step (S 184) for processing the audio signal separated by the separating step (S 183)

A binaural flag signal indicating that a binaural microphone (3) to be worn on the ear of the photographer (300) is used as the microphone for collecting the ambient sound is recorded on the recording medium (44) A step (S 186) of acquiring the binaural flag signal when acquiring the binaural flag signal in the flag information obtaining step (S 186), and a step of acquiring the binaural flag signal And a crosstalk canceling step (S 188) of processing the audio signal so as to cancel a crosstalk signal generated when the sound signal processed by the speaker (53, 54) is sounded,

The crosstalk cancellation step (S188) is a step of convolving the filter characteristics obtained in advance with respect to the voice signal, and the filter characteristic is a sound obtained by converting the measurement signal into a sound collected by a pair of microphones mounted on the cylindrical structure Wherein the filter characteristic is a filter characteristic based on a head related transfer function measured using a signal.

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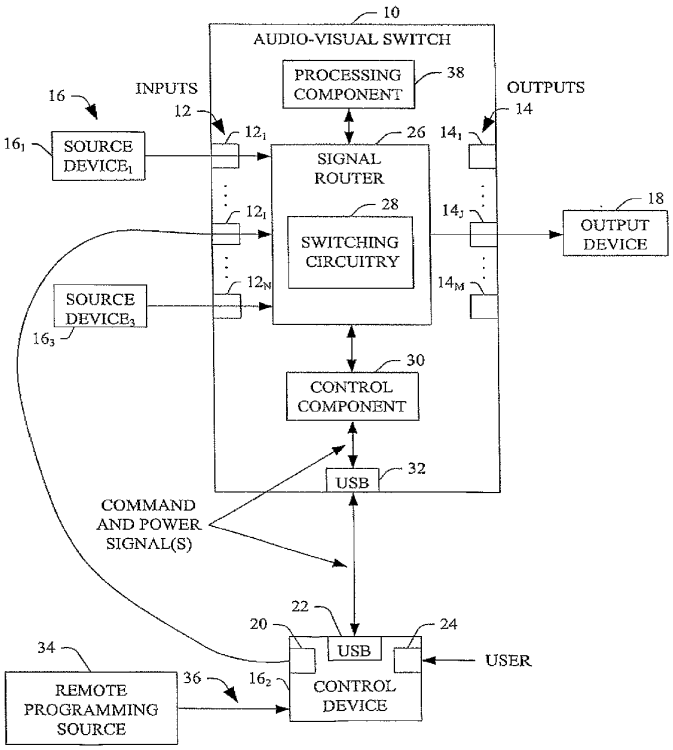
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(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LV, LY, MA, MD, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW

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[Continued on next page]

(54) Title: USB-ENABLED AUDIO-VIDEO SWITCH



(57) Abstract: A telemedicine system (8) includes an audio-video switch (10) that receives at least two different signals and conveys at least one of the at least two different signals to an output device (18) that presents it to a user. A control device (162) monitors a position of the audio-video switch (10) and automatically controls the position to select which of the at least two different signals is passed through the audio-video switch (10) to the output device (18) in response to a signal from a remote source (34). Signals received from the remote source (34) include medically related audio video message signals, which are input to the audio-video switch (10), and a control signal that causes the audio-video switch (10) to select between passing the medically related audio-video message signals and other received signals to the output device (18).

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FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, **MR,** NE, SN, TD, TG).

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B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04N G06F

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)
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	EP 1 126 715 A2 (KATZ RONALD ALAN [US] TELEBUYER LLC [US]) 22 August 2001 (2001-08-22) page 3, paragraph 13 page 9, paragraph 70 page 16, paragraph 134 - page 17, paragraph 151; figures 1-5	1-26
X	US 2005/052527 A1 (REMY CHRISTOPHE [US] ET AL) 10 March 2005 (2005-03-10) page 3, paragraph 30 - paragraph 32; figure 3 page 5, paragraph 41 - paragraph 42; figures 7,8	1-26

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[X] Further documents are listed in the continuation of Box C

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

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Date of the actual completion of the international search	Date of mailing of the international search report
3 Apri l 2007	13/04/2007

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P B 5818 Patentlaan 2 NL- 2280 HV Rijεwijk Tel (+31-70) 340-2040,Tx 31 651 epo nl, Fax- (+31-70) 340-3016	Schinnerl , Arnold

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C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 6 717 609 B2 (SUGIMOTO HIDEO [JP] ET AL) 6 April 2004 (2004-04-06) column 7, line 19 - column 8, line 42 abstract; figures 1-5	1-26
X	US 5 553 609 A (CHEN YAOBIN [US] ET AL) 10 September 1996 (1996-09-10) column 9, line 57 - column 12, line 53; figures 5-9	1,23,26
A	EP 1 343 326 A1 (MACROSYSTEM DIGITAL VIDEO A6 [DE]). 10 September 2003 (2003-09-10) abstract; figure 1 column 6, line 22 - line 58	1-26

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/US2007/060620

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 1126715 A2	22-08-2001	NONE	
US 2005052527 A1	10-03-2005	US 2006119701 A1	08-06-2006
US 6717609 B2	06-04-2004	DE 10101064 A1	19-07-2001
		JP 2001197485 A	19-07-2001
		US 2003197781 A1	23-10-2003
US 5553609 A	10-09-1996	AU 4980596 A	27-08-1996
		WO 9624284 A1	15-08-1996
EP 1343326 A1	10-09-2003	AT 275799 T	15-09-2004
		DE 50200955 D1	14-10-2004
		US 2003193577 A1	16-10-2003

Note: This international search report was established in addition to the report duly established by the competent International Searching Authority specified by the applicant. It is published for information only and has no legal status for the purposes of the PCT procedure (for example, in the computation of time limits).

INTERNATIONAL SEARCH REPORT

International application No.
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A. CLASSIFICATION OF SUBJECT MATTER
IPC: **H04N 7/18**(2006.01**),9/47**(2006.01)

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Minimum documentation searched (classification system followed by classification symbols)
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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)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	US 5,553,609 (CHEN et al.) 10 September 1996, entire document	1-3, 5, 7-11, 13-24 --- 4, 6, 12, 25-26
Y	US 2008/005421 A1 (CHANG et al.) 03 January 2008; entire document	4, 6, 12, 25-26

☐ 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

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

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"&" document member of the same patent family

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Date of mailing of the international search report
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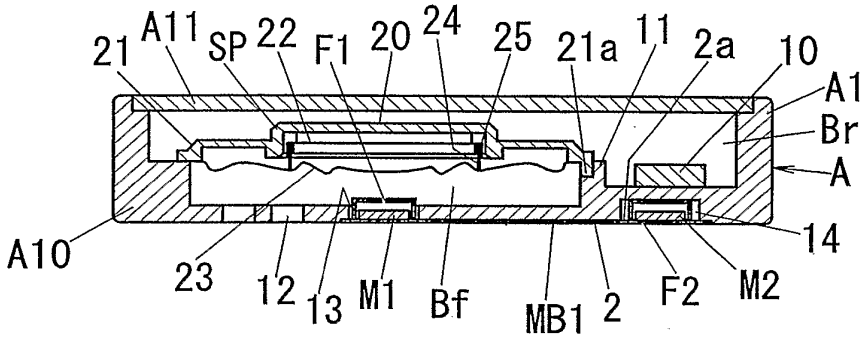
(74) Agents: **NISHIKAWA, Yoshikiyo** et al.; Hokuto Patent Attorneys Office Umeda-Daiichiseimei Bldg. 5th Floor, 12-17, Umeda 1-chome, Kita-ku, Osaka-shi, Osaka 5300001 (JP).

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(54) **Title:** INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME



(57) **Abstract:** A compact intercom device is provided, which efficiently prevents howling phenomenon. This intercom device includes a housing; a speaker accommodated in the housing such that a front air chamber is defined at a front side of the speaker and a rear air chamber is defined as a rear side of the speaker, a first microphone disposed such that its sound receiving surface faces the front air chamber, a second microphone disposed such that its sound receiving surface faces outside of the housing; and a signal processing unit. When an audio output of the speaker is picked up by the second microphone, the signal processing unit removes a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

WO 2008/050583 A1

DESCRIPTION

INTERCOM DEVICE AND WIRING SYSTEM USING THE SAME

TECHNICAL FIELD

The present invention relates to an intercom device suitable for an interphone system or the like, and a wiring system using the same.

BACKGROUND ART

Conventionally, an interphone system has been widely used as short-distance conversation means between separated rooms or between an entrance and a room of a building or a house. A typical interphone system for household use is mainly formed with an indoor intercom device installed on a wall surface of the room, and an outdoor intercom device installed on a wall surface of the entrance, and connected to the indoor intercom device through a transmission line. For example, the outdoor intercom device has a microphone for receiving a visitor's voice, and a speaker for outputting a dweller's voice. By operating the outdoor intercom device, the visitor can have a conversation with the dweller using the indoor intercom device.

In this kind of intercom device, since the speaker is located near the microphone, there is an inconvenience that howling phenomenon easily occurs when the audio output of the speaker is picked up by the microphone. To solve this inconvenience, various measures or methods for preventing the howling phenomenon have been proposed.

For example, Japanese Patent Publication No. 2607257 discloses a loudspeaker communication device **300**, which is suitable for a handsfree telephone. As shown in FIG. 35A, this communication device **300** has a speaker **SP**, and a pair of microphones (**M1**, **M2**). In addition, to prevent the howling phenomenon caused when an audio output of the speaker **SP** is picked up by the microphone (**M1**, **M2**), the communication device **300** also includes a pair of amplifying circuits (**310**, **320**), a level-control amplifying circuit **330**, a

delay circuit **340**, and a differential amplifying circuit **350**, as shown in FIG. 35B. The delay circuit **340** delays an output of the microphone **M1** located closer to the speaker **SP** by a delay time, which is determined according to sound velocity and a difference between a distance **d1** between the microphone **M1** and the speaker **SP** and a distance **d2** between the microphone **M2** and the speaker **SP**. The level-control amplifying circuit **330** controls the output level of the microphone **M2** such that an amplification factor is **d2/d1**. In other words, the output level of the microphone **M2** is controlled so as to be substantially equal to the output level of the microphone **M1** with respect to the signal component corresponding to the audio output of the speaker **SP**. The differential amplifying circuit **350** receives an output of the delay circuit **340** and an output of the level-control amplifying circuit **330**, and provides an output signal corresponding to a difference between these outputs.

According to this communication device, the signal component corresponding to the audio output of the speaker **SP** is cancelled from the output of the microphone (**M1**, **M2**) by the differential amplifying circuit **350**. Therefore, it is possible to achieve a comfortable communication, while preventing the howling phenomenon.

However, as shown in FIG. 35A, since each of the microphones (M1, M2) is spaced from the speaker SP by a relatively large distance, there is another problem that the device itself becomes large in size. In addition, since the intercom device is usually installed on the wall surface or the like, it is strongly desired that the intercom device has a thin thickness. On the other hand, when the speaker is accommodated in an excessively thin housing, a volume of a rear air chamber, which is defined as a space of a rear side of the speaker, is reduced, so that a radiating sound pressure of the speaker lowers and a minimum resonance frequency of the speaker shifts to a higher frequency level. As a result, there is a fear that deteriorations in sound quality and electro-acoustic conversion efficiency of the speaker occur.

Thus, the conventional intercom device still has plenty of room for

improvement in downsizing of the device itself, sound quality, electro-acoustic conversion efficiency as well as the prevention of howling phenomenon.

SUMMARY OF THE INVENTION

Therefore, a primary concern of the present invention is to provide a compact intercom device, which has the capability of preventing howling phenomenon, and achieving an improvement in sound quality and efficiency.

That is, the intercom device of the present invention is characterized by comprising:

a housing;

a speaker accommodated in the housing such that a front air chamber is defined as a space enclosed by a front surface of the speaker and an inner surface of the housing, and a rear air chamber is defined as a space enclosed by a rear surface of the speaker and an inner surface of the housing;

a first microphone disposed such that its sound receiving surface faces the front air chamber;

a second microphone disposed such that its sound receiving surface faces outside of the housing; and

a signal processing unit configured to, when an audio output of the speaker is picked up by the second microphone, remove a signal component corresponding to the audio output of the speaker from an output signal of the second microphone by use of an output signal of the first microphone.

According to the invention, since the first microphone easily and efficiently collects the audio information output from the speaker, howling phenomenon can be efficiently prevented by the signal processing unit. In addition, a distance between the first microphone and the speaker becomes short, thereby allowing downsizing of the intercom device. Furthermore, since the sound receiving surface of the second microphone faces the outside of the housing, an acoustic coupling between the speaker and the second microphone reduces, and the second microphone becomes hard to pick up the audio output of the

speaker. As a result, the second microphone can be located closer to the speaker. Thus, it is possible to provide a compact intercom device capable of preventing the howling phenomenon.

From the viewpoint of more efficiently preventing the howling phenomenon, it is preferred that the rear air chamber is the space hermetically-sealed by the rear surface of the speaker and the inner surface of the housing. In addition, it is preferred that the first microphone is arranged such that the sound receiving surface of the first microphone faces a diaphragm of the speaker. Moreover, when the first microphone and the second microphone are mounted on a same circuit board, it is preferred that the circuit board is mounted on an outside surface of the housing such that the sound receiving surface of the first microphone faces the front air chamber and the sound receiving surface of the second microphone faces the outside of the housing.

On the other hand, from the viewpoint of further downsizing the intercom device, the second microphone is preferably located adjacent to the speaker. In addition, it is preferred that the first and second microphones are mounted on a single board.

From the viewpoint of improving sound quality and electro-acoustic conversion efficiency of the speaker, it is preferred that the intercom device has an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end. Alternatively, the intercom device may have a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end. In addition, it is preferred that the acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of the speaker is needed.

Furthermore, it is preferred that the acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant frequency of the speaker that is supposed to be mounted on an infinite baffle and a lowest

resonant frequency of the speaker mounted on the rear air chamber with the absence of the acoustic tube. The acoustic tube can be formed along an inner surface of the rear air chamber. If necessary, a sound absorbing material may be disposed at the opening of the acoustic tube.

A further concern of the present invention is to provide a wiring system using the intercom device described above. That is, the wiring system comprises at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of the building structure, and connected to the transmission line. The intercom device has a first connector detachably connectable to a second connector formed in each of the base units. This wiring system is characterized in that when the first connector is connected to the second connector of one of the base units, the intercom device functions such that the information signals provided through the transmission line are output as audio information from the speaker unit, and an output of the signal processing unit is sent out through the transmission line.

By the way, in the intercom device installed on the wall surface of the house, one provided with a liquid crystal display or the like for displaying not only the audio information but also visual information such as images comes into practical use recently. Once such an intercom device is installed on the wall surface, not only electric wiring work but also mounting/repair work to the wall surface are required in a case where a layout of the intercom device has to be changed. However, it is not easy for a general user himself/herself to do the work. In addition, a function of the conventional intercom device is completed by itself, and a new-type intercom device is required in a case where it is desired to add another function. In such a case, not only a purchase of the new-type intercom device but also the above-described mounting/repair work becomes necessary, so that this is a great burden in cost for the user. The wiring system of the present invention is effect. According to the wiring system of the present

invention, since the intercom device is detachably connectable to the base unit, there advantages that the wiring system allows great flexibility for the change of layout of the intercom device, and makes it easy to replace the intercom device without cumbersome repairing work. Thus, the present invention provides the wiring system capable of solving the problems described above.

As a preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units, the function unit having a third connector detachably connectable to one of the second connector of the base unit, and an extended connector formed in the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, or when the first connector of the intercom device is connected to the second connector of the base unit, and the third connector of the function unit is connected to the extended connector of the intercom device, the function unit provides at least one of functions for supplying electric power provided through the transmission line, outputting information signals provided through the transmission line, and sending out information signals through the transmission line. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

As another preferred embodiment of the wiring system of the present invention, the wiring system further comprises a function unit adapted in use to be connected to the transmission line through one of the base units. The function unit has a third connector detachably connectable to the second connector of the base unit and an extended connector detachably connectable to the first connector of the intercom device. When the third connector of the function unit is connected to the second connector of the base unit, and the extended connector of the function unit is connected to the first connector of the

intercom device, the function unit provides at least one of functions for supplying electric power provided from the transmission line through the base unit, outputting information signals provided from the transmission line through the base unit, and sending out information signals to the transmission line through the base unit, and the intercom device operates such that the information signals provided from the transmission line through the base unit and the function unit are output as audio information from the speaker unit, and the output of the signal processing unit is sent out to the transmission line through the base unit and the function unit. At this time, it is particularly preferred that the first connector and the third connector are stylized in shape and arrangement, and the second connector and the extended connector are stylized in shape and arrangement corresponding to the stylization of the first connector and the third connector.

In the wiring system described above, it is preferred that the first and second connectors provide an electric power transmission between the base unit and the intercom device by means of electromagnetic coupling, and/or the first and second connectors provide a signal transmission between the base unit and the intercom device by means of optical coupling.

Further characteristics of the present invention and advantages brought thereby will be clearly understood from the best mode for carrying out the invention described below.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a perspective view of an intercom device according to a first embodiment of the present invention;

FIG. 2 is a cross-sectional view of the intercom device;

FIG. 3 is a circuit diagram of a voice processing unit of the intercom device;

FIG. 4 is a cross-sectional view of a microphone board of the intercom device;

FIG. 5 is a cross-sectional view of a bare chip employed in the microphone board;

FIG. 6A is a simplified plan view of the microphone board, and FIG. 6B is a simplified circuit diagram of the microphone board;

FIG. 7 is a circuit diagram of an impedance converting circuit of the microphone board;

FIG. 8 is a circuit diagram of a signal processing unit of the intercom device;

FIG. 9A and FIG. 9B are diagrams showing signal waveforms in the signal processing unit;

FIG. 10A and FIG. 10B are diagrams showing further signal waveforms in the signal processing unit;

FIG. 11A and FIG. 11B are diagrams showing another signal waveforms in the signal processing unit;

FIGS. 12A to 12C are diagrams showing still another signal waveforms in the signal processing unit;

FIG. 13 is a cross-sectional view of an intercom device according to a second embodiment of the present invention;

FIGS. 14A to 14C are cross-sectional and plan views of another housing of the intercom device according to the second embodiment;

FIG. 15 is a graph showing frequency characteristics of the cancelled amount of voice components of the speaker;

FIG. 16 is a graph showing frequency characteristics of the radiating sound pressure of the speaker;

FIG. 17 is a cross-sectional view of an intercom device using an ideal baffle plate;

FIG. 18 is a graph showing sound pressure characteristics of the intercom device having an acoustic tube and the intercom device having no acoustic tube;

FIG. 19 is a simplified cross-sectional view showing a modification of the housing having the acoustic tube;

FIG. 20 is a simplified cross-sectional view of a housing having a plurality of acoustic tubes;

FIG. 21 is a plan view of a housing having a sound absorbing material;

FIG. 22 is a schematic diagram of a wiring system using an intercom device according to a third embodiment of the present invention;

FIG. 23 is a schematic circuit diagram of a base unit of the wiring system;

FIG. 24 is a rear perspective view of the base unit;

FIG. 25A is a perspective view showing a gate housing and a main housing of the base unit, and FIG. 25B is a plan view of a module port of the gate housing;

FIG. 26 is a plan view of an attachment plate for mounting the base unit on a switch box;

FIG. 27 is a schematic circuit diagram of the function unit;

FIG. 28 is a perspective view of the intercom device for the wiring system;

FIG. 29 is a schematic circuit diagram of the intercom device for the wiring system;

FIG. 30 is a perspective view of the intercom device connected to the function unit or the base unit;

FIG. 31 is an exploded perspective view of the base unit;

FIG. 32A is a front view of the function unit, FIG. 31B is a side view of the function unit, FIG. 32C is an exploded side view of the function unit, and FIG. 32D is a perspective view of a joining member;

FIG. 33 is a perspective view illustrating how to mechanically couple the function unit to the base unit;

FIG. 34 is a schematic circuit diagram of an intercom device for a power line carrier type wiring system; and

FIG. 35A is a schematic perspective view of a conventional loudspeaker communication device, and FIG. 35B is a partial block diagram of the loudspeaker communication device.

BEST MODE FOR CARRYING OUT THE INVENTION

Referring to the attached drawings, an intercom device of the present invention and a wiring system using the intercom device are explained in detail

according to preferred embodiments.

(First Embodiment)

As shown in FIGS. 1 and 2, an intercom device **A** according to a first embodiment of the present invention includes a housing **A1**, a speaker **SP**, a microphone board **MB1**, and a voice processing unit **10**. The housing **A1** consists of a body **A10** having an opening at its back face, and a cover **A11** for closing the opening of the body **A10**, and accommodates the speaker **SP** and the voice processing unit **10** therein. The housing **A10** has a plurality of sound holes **12** formed in a front face of the body **A10**. The speaker **SP** accommodated in the housing **A1** provides a voice output through the sound holes **12** to outside of the housing **A1**. The microphone board **MB1** has a sound hole **F2** that picks up a voice emitted from a user of the intercom device **A**. In operation, when a conversation switch **SW1** is operated to the ON position, the intercom device **A** sends the picked-up voice to another intercom device **A** and receives signals from another intercom device **A** via an information line **Ls**.

As shown in FIG. 2, the speaker **SP** includes a cylindrical yoke **20** that is open at one end, and a round support **21** that extends radially outward from the open end of the yoke **20**. The yoke **20** is formed of an iron-containing material, such as a cold rolling steel plate (SPCC, SPCEN) or an electromagnetic soft iron (SUY), and has a thickness of about 0.8 mm. A columnar permanent magnet **22** (having a residual flux density of, for example, 1.39 T to 1.43 T) formed of, for example, neodymium is placed in the yoke **20**. An edge surface of the round support **21** is fixed on an edge portion on an outer peripheral side of a dome-shaped diaphragm **23**.

The diaphragm **23** is formed of a thermoplastic material, such as PET (PolyEthylene Terephthalate) or PEI (Polyetherimide), and has a thickness of, for example, 12 μm to 50 μm. A cylindrical bobbin **24** is fixed to the back of the diaphragm **23**, and a voice coil **25** is provided at the rear end of the bobbin **24**. The voice coil **25** is located at the open end of the yoke **20**, and the bobbin **24**

and the voice coil **25** are allowed to freely move back and forth (in the vertical direction in FIG. 2) in the vicinity of the open end of the yoke **20**. The voice coil **25** is formed by winding a polyurethane copper wire (having a diameter of 0.05 mm, for example) around a paper sleeve of craft paper.

When the polyurethane copper wire of the voice coil **25** receives a voice signal, an electromagnetic power is generated in the voice coil **25** by electric current of the voice signal and a magnetic field of the permanent magnet **22**, thereby to vibrate the bobbin **24**, along with the diaphragm **23**, forward and backward. As a result, a sound corresponding to the voice signal is emitted from the diaphragm **23**. That is to say, the speaker **SP** of this embodiment is an electrodynamic speaker. The speaker has, for example, a diameter of 20 mm to 25 mm, and a thickness of about 4.5 mm.

A rib **11** is formed on an inner surface of the front wall of the housing **A1**, which is opposed to the diaphragm **23**, and is in contact with an end face of a convex portion **21a** that protrudes frontward from an outer peripheral portion of the round support **21**. Thus, the speaker **SP** is fixed to the housing **A1** such that the diaphragm **23** faces the inner surface of the front wall of the housing **A1**.

With the speaker **SP** thus fixed within the housing **A1**, a front air chamber **Bf** is formed as a space enclosed by the inner surface of the front wall of the housing **A1** and a front surface (on which the diaphragm **23** is located) of the speaker **SP**, and a rear air chamber **Br** is formed as a space enclosed by an inner surface of the rear and side walls of the housing **A1** and the rear surface (on which the yoke **20** is located) of the speaker **SP**. The front air chamber **Bf** communicates with outside of the housing **A1** through the sound holes **12** formed in the front wall of the housing **A1**. The rear air chamber **Br** is insulated from (or does not communicate with) the front air chamber **Bf** since the end portion of the support **21** and the rib **11** are attached firmly to each other. Furthermore, the cover **A11** is tightly fitted in the opening formed in the rear wall of the body **A10**, so that the rear air chamber **Br** provides a

hermetically-sealed space that is insulated also from the outside of the housing **A1**. A gasket may be provided between the end portion (or the convex portion **21a**) of the support **21** and the rib **11** so as to increase the degree of adhesion between the support **21** and the rib **11**.

As shown in FIG. 3, the voice processing unit **10** is formed by an integral circuit (IC), which includes a communication unit **10a**, echo canceling units (**10b**, **10c**), an amplification unit **10d** and a signal processing unit **10e**. In operation, the communication unit **10a** receives a voice signal transmitted from an intercom device **A** installed, for example, in another room through the information line **Ls**. The received voice signal is transmitted through the echo canceling unit **10b** to the amplification unit **10d**, in which the signal is amplified, and then output from the speaker **SP**. When the conversation switch **SW1** is operated to the ON position, the voice communication function of the intercom device **A** becomes available. In this condition, voice signals respectively received from a microphone **M1** (first microphone) and a microphone **M2** (second microphone) on the microphone board **MB1** are processed by the signal processing unit **10e**. In the signal processing unit **10e**, each received signal is processed in the manner as described later, and the resulting signal passes through the echo canceling unit **10c**. Then, upon reception of the resulting signal from the echo canceling unit **10c**, the communication unit **10a** transmits the resulting signal through the information line **Ls** to the intercom device **A** installed in, for example, another room. In this manner, the intercom device **A** serves as an interphone that permits two-way voice communications between the rooms. An electric power for the intercom device **A** may be supplied from a wall socket disposed in the vicinity of the location where the intercom device **A** is installed. Alternatively, the electric power may be supplied through the information line **Ls**, or may be supplied from an internal battery accommodated in the intercom device **A**.

As shown in FIG. 4, the microphone board **MB1** has a module board **2** and the microphones **M1** and **M2**, which are mounted on a (top) surface **2a** of the

module board **2**. The microphone **M1** is mainly formed with a bare chip **BC1**, an IC chip **Ka1**, and a shield case **SC1**. Each of the bare chip **BC1**, the IC chip **Ka1**, and a wiring pattern (not shown) on the module board **2** are connected to each other with wires **W** (wire bonding). The shield case **SC1** covers the bare chip **BC1** and the IC chip **Ka1**. The microphone **M2** has the same construction as that of the microphone **M1**. That is to say, the microphone **M2** is formed with a bare chip **BC2**, an IC chip **Ka2**, and a shield case **SC2** mounted on the module board **2** that covers the bare chip **BC2** and the IC chip **Ka2**. The microphones (**M1**, **M2**) are preferred to be formed on a silicon substrate by a micro structure manufacturing process. That is, it is preferred that the microphones (**M1**, **M2**) are IC chips of Micro Electro Mechanical System (MEMS).

Each of the bare chips **BC1** and **BC2**, which will be generally referred to as "bare chip **BC**", provides a condenser-type silicon microphone, as shown in FIG. 5. More specifically, the bare chip BC includes a silicon thin film **1d** that is formed on a (top) surface of a silicon substrate **1b** so as to cover a cavity **1c** formed through the silicon substrate **1b**. An electrode **1f** is formed above the silicon thin film **1d** with an air gap **1e** interposed therebetween, and two pads **1g** are provided on the surface of the silicon substrate **1b** for producing an output in the form of a voice signal. When the silicon thin film **1d** is vibrated by a sound wave from outside, such as a human voice, a capacitance between the silicon thin film **1d** and the electrode **1f** changes, namely, the amount of electric charge changes. With the change in the electric charge amount, electric current proportional to the received sound wave flows from the two pads **1g**. Each of these bare chips **BC1** and **BC2** is fixed by die-bonding the silicon substrate **1b** on the module board **2**, as shown in FIG. 4. The silicon thin film **1d** of the bare chip **BC2** faces the sound hole **F2** formed in the module board **2**.

As shown in FIG. 4, a sound receiving surface of the microphone **M1** is positioned on a surface having the sound hole **F1** of the shield case **SC1**. On the other hand, a sound receiving surface of the microphone **M2** is positioned

on a surface having the sound hole **F2** of the module board **2**. Thus, the sound receiving surface of the microphone **M1** is to receive sound from an upper side of FIG. 4, and the sound receiving surface of the microphone **M2** is to receive sound from an opposite side, i.e., the lower side of FIG. 4. In brief, as shown in FIG. 2, the module board **2** is mounted to the housing **A1** such that the sound receiving surface of the microphone **M1** faces the diaphragm **23** of the speaker **SP**, and the sound receiving surface of the microphone **M2** faces outside of the housing **A1**. Since both of the microphones (**M1**, **M2**) are mounted on the same surface **2a** of the module board **2**, the thickness of the microphone board **MB1** can be advantageously reduced.

FIG. 6A is a simplified plan view of the module board **2**. The module board **2** has a substantially T-shape, which is composed of a rectangular portion **2f**, on which the microphone **M1** is mounted, a rectangular portion **2g**, on which the microphone **M2** is mounted, and a connecting portion **2h** that connects the rectangular portions **2f** and **2g**. The rectangular portion **2g** is formed to be larger than the rectangular portion **2f**. The rectangular portion **2g** has a negative power supply pad **P1**, a positive power supply pad **P2**, and output pads (**P3**, **P4**) on an edge portion of the rectangular portion **2g** such that each of the pads (**P1**, **P2**, **P3**, **P4**) forms a line along an end face of the edge portion of the rectangular portion **2g**.

As shown in FIG. 6B, the negative power supply pad **P1** is connected to a negative terminal of an external power supply, and the positive power supply pad **P2** is connected to a positive terminal of the external power supply. The electric power received from the pads (**P1**, **P2**) is supplied to the microphones (**M1**, **M2**) through the wiring patterns on the module board **2**. A voice signal corresponding to the voice picked up by the microphone **M1** is output from the output pad **P3** via the wiring pattern on the module board **2**. On the other hand, a voice signal corresponding to the voice picked up by the microphone **M2** is output from the output pad **P4** via the wiring pattern on the module board **2**. The negative power supply pad **P1** also serves as a ground pad that establishes

a ground for the voice signals delivered from the output pads (**P3**, **P4**).

As described above, electric powers needed to activate the microphones (**M1**, **M2**) are supplied from the power supply via the common negative and positive power supply pads (**P1**, **P2**). The negative power supply pad **P1** also serves as the ground pad for the outputs of microphones (**M1**, **M2**). Therefore, the number of pads used in the microphone board **MB1** can be reduced, and the construction of the microphone board **MB1** can be simplified.

Next, an operation of the microphone board MB1 will be explained. First, electric currents flowing from the bare chips **BC1** and **BC2** corresponding to the received sound signal are subjected to impedance-conversion, and converted to voltage signals by the IC chips **Ka1** and **Ka2**. Then, the voltage signals are output as voice signals from the output pads (**P3**, **P4**), respectively.

For example, each of the IC chip **Ka1** and the IC chip **Ka2**, which will be generally referred to as "IC chip **Ka**", has a circuit configuration, as shown in FIG. 7. The IC chip **Ka** has a constant-voltage circuit **Kb** formed of a chip IC that converts a power supply voltage +V (e.g., 5V) supplied from the power supply pads (**P1**, **P2**) to a constant voltage Vr (e.g., 12V). The constant-voltage circuit **Kb** applies the constant voltage Vr to a series circuit of a resistance **R11** and the bare chip **BC**. A connection midpoint of the resistance **R11** and the bare chip **BC** is connected to a gate terminal of a junction-type J-FET **S11** through a capacitor **C11**. A drain terminal of the J-FET **S11** is connected to an operating voltage +V, and a source terminal is connected through a resistance **R12** to the negative terminal of the power supply. Herein, the J-FET **S11** is intended for converting electrical impedance, and the voltage of the source terminal of the J-FET **S11** produces an output as the voice signal. It is to be understood that the impedance converting circuit of the IC chip **Ka** is not limited to the above-described configuration, and may be replaced by a circuit having a function of a source follower circuit by an operation amplifier. For example, an amplifying circuit for the voice signal may be provided in the IC chip **Ka**, if necessary.

In the microphone board **MB1**, on which signal transmission and supply of electric power are performed via the wiring patterns on the module board **2** as described above, the signal lines and the power-supply lines can be arranged with high efficiency, and attached to the outer surface of the housing **A1**, as shown in FIG. 1. In this embodiment, as shown in FIG. 1, the surface **2a** of the module board **2** is attached along an outer surface of the front wall of the housing **A1**, and the microphone **M1** is inserted in an opening **13** formed in the front wall of the housing **A1** such that the sound receiving surface of the microphone **M1** faces the front air chamber **Bf**. The diaphragm **23** faces the microphone **M1** through the sound hole **F1** formed in the shield case **SC1**. With this arrangement, the microphone **M1** has a high directivity to the voice emitted from the speaker **SP** via the sound hole **F1**, so that the voice emitted from the speaker **SP** can be efficiently picked up by the microphone **M1**. The microphone **M2** fits in a recess **14** formed in the front wall of the housing **A1**, and faces the outside (or the front) of the housing **A1** through the sound hole **F2** formed in the module board **2**. In this regard, a direction where the second microphone **M2** receives a voice from the outside of the housing **A1** is substantially equal to the direction of emitting a voice from the speaker **SP**. Thus, the microphone **M2** has a high directivity to the voice of a talker who is located in front of the intercom device **A**. When **X1**, **X2** represent distances from the center of the speaker **SP** to the centers of microphones (**M1**, **M2**), respectively, the relationship between the distances (**X1**, **X2**) is expressed as **X1** $<$**X2**.

Since the rear air chamber **Br** facing the rear surface of the speaker **SP** is hermetically-sealed in the housing **A1** as shown in FIG. 2, the voice emitted from the rear surface of the speaker **SP** (rear surface of the diaphragm **23**) is less likely to leak from the rear air chamber **Br**, so that an acoustic coupling between the speaker **SP** and the microphone **M2** is lowered. BY the way, the voice emitted from the rear surface of the speaker **SP** is reversed in phase with respect to the voice emitted from the front surface of the speaker **SP** (front surface of the

diaphragm **23**). Therefore, if the voice emitted from the rear surface of the speaker **SP** goes around to the front of the speaker **SP**, these voices emitted from the front and rear surfaces of the speaker **SP** are cancelled with each other. As a result, the radiating sound pressure of the speaker **SP** is lowered, and it becomes difficult for the talker located in front of the intercom device **A** to hear the voice (sound) emitted from the speaker **SP**. However, since the intercom device of this embodiment is formed to reduce the leaking of the voice emitted from the rear surface of the speaker **SP** to the outside of the housing **A1**, as described above, it is possible to prevent such a reduction in the radiating sound pressure, which is caused by the voice emitted from the rear surface of the speaker **SP** going around to the front of the speaker **SP**.

As shown in FIG. 2, since the recess **14**, in which the microphone **M2** is inserted, is an insulated space not communicated with the rear air chamber **Br**, the microphone **M2** is even less likely to pick up the voice (or sound) emitted from the speaker **SP**. Therefore, the acoustic coupling between the speaker **SP** and the microphone **M2** can be further reduced. With the above arrangement, the microphones (**M1**, **M2**) are adapted to separately pick up the voice emitted from the speaker **SP** and the voice of (generated by) the talker, respectively.

Furthermore, if the microphone board **MB1** is placed in the housing **A1**, it is difficult to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**. In this embodiment, however, the microphone board **MB1** is attached to the outer surface of the housing **A1**. Therefore, it is possible to maintain the spatial insulation between the front air chamber **Bf** and the rear air chamber **Br**.

As described below, the intercom device of this embodiment has a configuration to prevent a howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**.

As shown in FIG. 8, the signal processing unit **10e** of the voice processing unit **10** includes an amplifying circuit **30** for amplifying the output of the microphone **M1**, a band-pass filter **31** for removing noises of frequencies other

than a voice-band (300 to 4000 Hz) from the output of the amplifying circuit **30**, and a delaying circuit **32** for delaying the output of the band-pass filter **31**. The signal processing unit **10e** further includes an amplifying circuit **33** for amplifying the output of the microphone **M2** while reversing the phase of the output of the microphone **M2** by 180 degrees, a band-pass filter **34** for removing noises of frequencies other than the voice-band (300 to 4000 Hz) from the output of the amplifying circuit **33**, and an adder circuit **35** for adding the outputs of the delaying circuit **32** and the band-pass filter **34**.

Referring to FIG. 8 to FIG. 12C, operations of the signal processing unit **10e** will be explained below. First, an operation of the signal processing unit **10e** when the voice emitted from the speaker **SP** is picked up by the microphones (**M1**, **M2**) will be explained. FIGS. 9A and 9B, 10A and 10B, 11A and 11B, and 12A to 12C show voice signal waveforms that appear at respective units of the signal processing unit **10e** when the microphones (**M1**, **M2**) respectively pick up the voice emitted from the speaker **SP**. As mentioned above, the distance **X1** from the center of the speaker **SP** to the center of the microphone **M1** is smaller than the distance **X2** from the center of the speaker **SP** to the center of the microphone **M2**. Accordingly, as shown in FIGS. 9A and 9B, when the microphones (**M1**, **M2**) pick up the voice emitted from the speaker **SP**, the amplitude of the output **Y21** of the microphone **M2** is smaller than that of the output **Y11** of the microphone **M1** due to the difference between the distances (**X1**, **X2**). In addition, the phase of the output **Y21** of the microphone **M2** is delayed with respect to that of the output **Y11** of the microphone **M1** by a delay time **Td** (=(**X2**-**X1**)/**Vs**, where **Vs** is sonic velocity), which is proportional to the difference between the distances (**X2**-**X1**).

The amplifying circuit **30** amplifies the output **Y11** to produce an output **Y12**, and the amplifying circuit **33** amplifies the output **Y21** while reversing the phase thereof by 180 degrees, to produce an output **Y22**. At this time, each of the amplifying circuits (**30**, **33**) performs level adjustment based on the difference between the distances (**X2**-**X1**), so that the output levels of the

microphones (**M1**, **M2**) relative to the voice emitted from the speaker **SP** become equal to each other, as shown in FIGS. 10A and 10B. In this embodiment, since an amplitude rate of the amplifying circuit **30** is set to substantially 1, the amplifying circuit **30** may be eliminated from the signal processing unit **10e**.

As shown in FIGS. 11A and 11B, the band-pass filters (**31**, **34**) remove frequencies other than the voice band from the outputs **Y12** and **Y22**, to produce outputs **Y13** and **Y23**, respectively.

The delaying circuit **32** can be formed by, for example, a time delay device or a CR phase delaying circuit. The delaying circuit **32** delays the output of the microphone **M1**, which is located closer to the speaker **SP**, by the above-described delay time **Td**, thereby producing an output **Y14** whose phase coincides with the phase of the output **Y23** of the band-pass filter **34**. Thus, it is possible to reduce noises contained in a voice signal to be transmitted.

Through the amplifying and delaying processes described above, a voice component emitted from the speaker **SP**, which is contained in the output **Y14**, and the voice component emitted from the speaker **SP**, which is contained in the output **Y23**, have the same amplitude and the reversed phase relative to each other, as shown in FIG. 12A and FIG. 12B. The adder circuit **35** adds the output **Y14** and the output **Y23** to cancel the voice component of the speaker **SP** from the output of the microphone **M2**. Thus, it is possible to produce an output **Ya** shown in FIG. 12C, in which the voice component from the speaker **SP** is removed or reduced. As described above, the signal processing unit **10e** removes the voice component of the speaker **SP** from the output signal **Y21** of the microphone **M2** by using the output signal **Y11** of the microphone **M1**.

On the other hand, an operation of the signal processing unit **10e** when the voice emitted from the talker **H** in front of the intercom device **A** is picked up by the microphones (**M1**, **M2**) will be described below. Since the microphone **M2** has a high directivity to the voice from the talker **H** located in front of the intercom device **A**, the amplitude of the output **Y21** of the microphone **M2** is larger than that of the output **Y11** of the microphone **M1**. Furthermore, since

the signal from the microphone **M1** is not amplified while the signal from the microphone **M2** is amplified in the amplifying circuit **33**, the voice component from the talker **H** contained in the output **Y23** is sufficiently larger than the voice component from the talker **H** contained in the output **Y14**. Thus, even after the above-described adding process is performed in the adder circuit **35**, the signal corresponding to the voice generated by the talker **H** remains in the output **Ya** with a sufficiently large amplitude.

In the manner described above, the voice component from the speaker **SP** is reduced in the output **Ya** of the adder circuit **35**, whereas the voice component generated by the talker **H** in front of the intercom device **A** remains in the output **Ya**. Thus, a relative difference in the output **Ya** between the voice component from the talker **H**, which is desired to be saved, and the voice component from the speaker **SP**, which is desired to be reduced, becomes larger. That is to say, even when the voice from the talker **H** and the voice from the speaker **SP** are simultaneously generated, only the voice component from the speaker **SP** is reduced while the voice component from the talker **H** remains sufficiently large. Accordingly, it is possible to prevent the howling phenomenon caused when the microphones (**M1**, **M2**) pick up the voice output of the speaker **SP**.

Since the microphone **M1** is disposed in the front air chamber **Bf** so as to face the diaphragm **23**, as shown in FIG. 2, the microphone **M1** efficiently picks up the voice emitted from the speaker **SP**. Therefore, the signal processing unit **10e** can efficiently achieve the howling phenomenon prevention function. Furthermore, since the sound receiving surface of the microphone **M2** faces outside of the housing **A1**, and the output direction of the speaker **SP** and the directivity of the microphone **M2** substantially conform to each other, the acoustic coupling between the speaker **SP** and the microphone **M2** is reduced. Thus, the microphone **M2** is less likely to pick up the voice emitted from the speaker **SP**, and can be placed close to the speaker **SP**, that is to say, close to the front air chamber **Bf**, which makes it possible to downsize the intercom device **A**.

On the other hand, the signal processing unit **10e** transmits the voice signal to the echo canceling unit **10c** through a digital-analog converter (not shown), and the echo canceling unit (**10b**, **10c**) shown in FIG. 3 perform the process as described below to further efficiently prevent the howling phenomenon.

First, the echo canceling unit **10c** retrieves the output of the echo canceling unit **10b** as a reference signal, and performs a suitable computation on the output of the signal processing unit **10e**, thereby canceling a voice signal, which goes around from the speaker **SP** to the microphones (**M1**, **M2**). On the other hand, the echo canceling unit **10b** also retrieves the output of the echo canceling unit **10c** as the reference signal, and performs a suitable computation on the output of the communication unit **10a**, thereby canceling a voice signal, which goes around from a speaker of an intercom device installed in another room to its microphones.

More specifically, the echo canceling units (**10b**, **10c**) control the amount of loss at variable loss means (not shown) provided in a loop circuit consisting of the speaker **SP**, the microphones (**M1**, **M2**), the signal processing unit **10e**, the echo canceling unit **10c**, the communication unit **10a**, the echo canceling unit **10b**, the amplifying unit **10d** and the speaker **SP**, so as to prevent the howling phenomenon by making the loop gain less than 1. Herein, one of the transmitted voice signal and the received voice signal, which has a smaller signal level, is regarded as not important, and a transmission loss of a variable loss circuit, which is inserted in a transmission path of the signal having the smaller level, is increased. In this embodiment, the intercom devices transmit signals via the information line **Ls**. However, the transmission of signals is not limited to a wired line. A non-wired transmission manner such as radio wave, optical communication system, and infrared radiation may be alternatively employed.

(Second Embodiment)

As shown in FIG. 13, an intercom device **A'** according to a second

embodiment of the present invention includes an acoustic tube **40**, which is provided in the rear air chamber **Br** having a small volume, so as to improve the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. The intercom device **A'** according to the second embodiment will be explained below.

The acoustic tube **40** has an opening at its one end and a closed end at the other end, and is provided in the rear air chamber **Br** to be along an inner wall surface(s) of the rear air chamber **Br**. By setting the resonance frequency of the acoustic tube **40** to an appropriate level, the lowest resonance frequency **fo** of the speaker **SP**, which is provided in the intercom device **A'**, is shifted to a lower frequency level, and the sound pressure level of the speaker **SP** increases, resulting in improvements of the sound quality and electro-acoustic conversion efficiency of the speaker **SP**. It is noted that except for the sound tube **40**, the construction of the intercom device **A'** shown in FIG. 13 is the same as that of the intercom device **A** shown in FIG. 2. Accordingly, details of the intercom device **A'** will be omitted herein.

The lowest resonance frequency **fo** of the speaker **SP** is generally determined by an equivalent mass (diaphragm, coil, additional mass of air) **Mo** of a vibration system of the speaker, stiffness **So** of an edge or the like supporting the vibration system, and stiffness **Sc** of air in the rear air chamber **Br**. That is, it is represented by an equation:

$$fo = \{1/(2\pi)\} \cdot \sqrt{\{(So+Sc)/Mo\}}$$

FIG. 14A through FIG. 14C show another housing **A2** of the intercom device **A'** according to a modification of the present embodiment. The housing **A2** accommodates the acoustic tube **40** therein. Effects of the sound tube **40** are explained below referring to the housing **A2** as an example. The housing **A2** essentially consists of a body **A20** having opening at its rear surface, and a cover **A21** for closing the opening of the body **A20**. In FIG. 14B, only the body **A20** is illustrated. As in the intercom device **A** of the first embodiment, the housing **A2** is capable of accommodating a speaker **SP**, a microphone board **MB1**, a

conversation switch **SW1**, and a voice processing unit **10**. The acoustic tube **40** is provided in the body **A20**.

The housing **A2** has two spaces **A2a** and **A2b** therein, which are insulated from each other. The space **A2a** has a lateral dimension of 30 mm, a vertical dimension of 40 mm, and a thickness of 8 mm, to accommodate the speaker **SP** therein. In the space **A2a**, columnar bosses **51** are formed at regularly spaced four locations on the inner surface of the front wall of the body **A20**, and screw holes **52** are respectively formed in the bosses **51**. The speaker **SP** is fixed to the inner surface of the front wall of the body **A20** by using mounting screws, which are thread engageable with the screw holes **52**. With the construction described above, a front air chamber **Bf** and a rear air chamber **Br** similar to those of the first embodiment (FIG. 2) are formed. The front air chamber **Br** is provided by a space enclosed by the inner surface of the front wall of the housing **A2** and the front surface side (diaphragm **23** side) of the speaker **SP**. The rear air chamber **Br** is provided by a space enclosed by the inner surface of the rear and side walls of the housing **A2** and the rear surface side (yoke **20** side) of the speaker **SP**. A plurality of sound holes **12** are formed in the front wall of the housing **A2**.

As shown in FIGS. 14A through 14C, the acoustic tube **40**, which has a rectangular shape in cross section, is formed along the inner walls of the rear air chamber **Br** at the back of the speaker **SP** over substantially three-quarter of the circumference of the rear air chamber **Br**. The acoustic tube **40** has an opening at one end **40a** and a closed end at the other end **40b**. The overall length **L** of the acoustic tube **40** is set to an odd multiple of one quarter of a wavelength, which is determined by the sonic velocity and a frequency **f** of which sound pressure level is desired to be increased. In this embodiment, the acoustic tube **40** has a length of one quarter of the wavelength determined by the frequency where an increase in sound pressure level of the audio output of the speaker **SP** is needed and the sonic velocity. In detail, since an open-end correction is performed on the acoustic tube **40**, where sonic velocity is set to C

and an open-end correction value is set to σ (=0.8**d**, wherein **d** is an open-end diameter of the acoustic tube **40**), the overall length is determined from an equation described below:

$$f = [C / \{4 (L+\sigma)\}]$$

In this embodiment, the overall length L of the acoustic tube **40** is set 95.2 mm and a cross-sectional area of the acoustic tube **40** is set to 4.2 mm^2 (equivalent to $\phi = 2.3$). By forming the acoustic tube **40** along the inner walls of the rear air chamber **Br**, it is possible to minimize a volume reduction of the rear air chamber **Br** caused by the formation of the acoustic tube **40**. In addition, with the acoustic tube **40** thus provided, the sound pressure level of the speaker **SP** can be increased in a desired frequency band (at around 700 Hz in this embodiment). As a result, it is possible to improve the sound quality and the electro-acoustic conversion efficiency of the speaker **SP**.

By utilizing the fact that input impedance becomes extremely small at the resonance frequency (i.e., a frequency at which the overall length of the tube is equal to an odd-number multiple of an approximately quarter (1/4) wavelength) of a closed tube, the acoustic tube **40** can reduce the sound leakage from the rear air chamber **Br** to the outside of the housing. That is, the sound waves radiated from the rear surface of the speaker **SP** is reflected toward the rear surface side of the speaker **SP**.

Referring to FIG. 15 through FIG. 18, effects of the acoustic tube **40** provided in the housing **A2** will be explained below in comparison with the intercom device **A** of the first embodiment, which has no acoustic tube in the housing **A1**, and an intercom device having an infinite baffle, which will be explained in detail below.

FIG. 15 shows frequency characteristics of the cancelled amount of the voice component from the speaker **SP** in the case of using the signal processing unit **10e** of intercom device **A** of the first embodiment, and FIG. 16 shows frequency characteristics of the radiating sound pressure measured in front of the speaker **SP**. FIG. 15 and FIG. 16 also show results obtained in the case

where the housing has no acoustic tube as in the first embodiment, and the case where an ideal baffle plate is used as the housing.

FIG. 17 shows a baffle plate **C** having an infinite size as an example of the above-mentioned ideal baffle plate, but the baffle plate **C** is not realistic for mounting a real speaker thereon. Thus, in this embodiment, a standard baffle as specified in JIS C 5532 is used to measure a frequency equivalent to a lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C**. The lowest resonant frequency of the speaker **SP** mounted on the standard baffle is equivalent to that of the speaker **SP** that is supposed to be mounted on the baffle plate **C** (an ideal baffle plate). The cancelled amount (**Y1a** in FIG. 15) in the case where the speaker **SP** and the microphone board **MB1** are mounted on the standard baffle to form an intercom device is kept at 10 dB or more in a frequency band of 100 Hz to 10000 Hz, and the radiating sound pressure characteristics (**Y2a** in FIG. 16) in the same case indicate that the lowest resonance frequency **fo1** of the speaker **SP** is 600 Hz, which is an equally ideal characteristic as provided by the speaker **SP** alone. A standard closed box as specified in JIS C 5532 may be used to measure a frequency equivalent to the lowest resonant frequency of the speaker **SP** that is supposed to be mounted on the baffle plate **C** having an infinite size (an ideal baffle plate) as shown in Fig. 17.

In the case where a rear air chamber having no acoustic tube **40** is used, as in the rear air chamber **Br** of the first embodiment, the cancelled amount (**Y1b** in FIG. 15) of the voice component from the speaker **SP** is kept at 10 dB or more in the frequency band of 100 Hz to 10000 Hz. Thus, substantially the same cancelled amount as that of the case where the standard baffle is used can be obtained. However, in the case where the rear air chamber **Br** is formed as an enclosed space, radiating sound pressure characteristics comparable to those of the case where the standard baffle is used can be obtained when the volume of the rear air chamber **Br** is sufficiently larger. On the other hand, when the volume of the rear air chamber **Br** is small, the mechanical equivalent stiffness

of the rear air chamber **Br** increases, and the lowest resonance frequency of the speaker **SP** becomes higher. As higher the lowest resonance frequency, the radiating sound pressure reduces, so that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** deteriorate. For example, the radiating sound pressure characteristics (**Y2b** in FIG. 16) of the case where the rear air chamber **Br** of the first embodiment is used indicate that the lowest resonance frequency **fo2** of the speaker **SP** is 1200 Hz, which is noticeably higher than the lowest resonance frequency obtained when the standard baffle is used. Furthermore, the sound pressure level in a frequency band below 800 Hz is lower by about 5 to 20 dB than that of the case where the standard baffle is used. It means that the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are deteriorated in this frequency band.

The radiating sound pressure characteristics (**Y2c** in FIG. 16) of the case where the volume of the rear air chamber **Br** is made three times larger than that of the rear air chamber **Br** of the first embodiment indicate that the lowest resonance frequency **fo3** of the speaker **SP** is 800 Hz, and the sound quality of the speaker **SP** is improved as compared with the case of using the housing **A1** of the first embodiment. However, the overall size of the housing increases with the increase in volume of the rear air chamber **Br**, which makes it difficult to downsize the intercom device. Thus, in this embodiment, by providing the acoustic tube **40** in the rear air chamber **Br** having a small volume, the sound quality and the electro-acoustic conversion efficiency of the speaker **SP** are improved, while making the intercom device smaller.

The overall length **L** of the acoustic tube **40** is set so that the resonance frequency of the acoustic tube **40** is located between the lowest resonance frequency **fo2** (=1200 Hz) of the case where the rear air chamber **Br** having no acoustic tube **40** is used, and the lowest resonance frequency **fo1** (=600 Hz) of the case where the standard baffle is used. By setting the overall length **L** of the acoustic tube **40**, as described above, the lowest resonance frequency **fo5** of

the speaker **SP** in the case where the housing **A2** having the acoustic tube **40** is used is set to about 700 Hz between **fo1** and **fo2**.

FIG. 18 shows frequency characteristics of the radiating sound pressure of the intercom device **A'**, which is provided with the housing **A2** and the sound tube **40** of the present embodiment, and the radiating sound pressure of the intercom device **A** of the first embodiment, which has no acoustic tube. The characteristics **Y3a** of the intercom device **A'** in FIG. 18 indicate that the sound pressure level in a frequency band not larger than 800 Hz is increased by about 5 to 10 dB, as compared with the characteristics **Y3b** of the intercom device **A**, and the lowest resonance frequency is reduced from **fo4** (=1200 Hz) to **fo5** (= about 700 Hz). That is, improvements in the speaker efficiency and the sound quality can be achieved. For example, the sound pressure level at around 700 Hz is increased by about 10 dB, and the improvement of the speaker efficiency brought by the increase of the sound pressure level corresponds to a 70% reduction in input current and a 90% reduction in input power of the speaker **SP**. Thus, the sound pressure level of the speaker **SP** increases around the lowest resonance frequency of the speaker **SP**. Therefore, by setting the overall length of the acoustic tube **40** to an appropriate length, and shifting the lowest resonance frequency of the speaker **SP** to a lower frequency side, it is possible to achieve improvements in sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As another preferred acoustic tube of the present invention, an acoustic tube **41** that communicates with the rear air chamber **Br** may be provided at the outside of a housing **A3**. For example, the acoustic tube **41** can be obtained by attaching a tubular member to into a hole (not shown) formed in the housing **A3**, as schematically shown in FIG. 19. When tubular member is made of urethane or the like, the overall length **L** of the acoustic tube **41** can be easily changed, if desired.

As a further modification of this embodiment, two or more acoustic tubes may be formed in the rear air chamber **Br**. As an example, three acoustic tubes

42, **43**, **44** can be formed in the rear air chamber **Br** of a housing **A4**, as schematically shown in FIG. 20. In this example, the acoustic tubes (**42**, **43**, **44**) have different lengths from each other. This means that they have mutually different resonance frequencies. Accordingly, the sound pressure level of the speaker **SP** is increased at a plurality of frequencies. By setting the overall lengths of the respective acoustic tubes (**42**, **43**, **44**) to appropriate values, it is possible to provide desired sound quality and electro-acoustic conversion efficiency of the speaker **SP**.

As shown by a dotted line in FIG. 21, a sound absorbing material **45** such as a nonwoven fabric may be provided at or in the vicinity of the open end **40a** of the acoustic tube **40**. By use of the sound absorbing material **45**, it is possible to provide an enhanced sound absorbing effect, and perform a fine adjustment of the resonance frequency of the acoustic tube **40**.

(Third Embodiment)

The present embodiment provides a wiring system using an intercom device **101** of the present invention. That is, this wiring system enables power and signal transmissions between electric devices spaced away from each other in a building structure.

As shown in FIG. 22, the wiring system comprises a power line **Lp** and an information line **Ls** installed in the building structure, which are connected to a commercial power source **AC** and the Internet network **NT** through a distribution board, switch boxes **102** embedded in wall surfaces of the building structure, base units **103** mounted on the switch boxes, and connected to the power line **Lp** and the information line **Ls**, and function units **104**, each of which is detachably connected to a desired one of the base units **103**, and has at least one of functions for supplying electric power from the power line **Lp**, outputting information from the information line **Ls**, and inputting information into the information line **Ls** in a connected state with the desired base unit. An intercom device **101** of the present invention, which is preferably provided by

the intercom device (**A**, **A'**) of the first or second embodiment described above, can be regarded as one of the function units **104**. In the present description, the wall surfaces of the building structure are not limited to surfaces of sidewalls standing between adjacent rooms. That is, the meaning of the wall surfaces includes outdoor and indoor wall surfaces, and the indoor wall surfaces comprise ceiling and floor surfaces as well as the sidewall surfaces. In the drawings, **MB** designates a main circuit breaker, **BB** designates a branch circuit breaker, and **GW** designates a gateway (e.g., a router or a built-in hub).

As shown in FIGS. 23 and 24, each of the base units **103** is formed with a gate housing **130** having terminals (**131a**, **132a**, **131b**, **132b**) connected to the power supply line **Lp** and the information line **Ls**, and a main housing **135** detachably connected to the function unit **104**. Theses housings can be made of a synthetic resin having electrical insulating characteristics (e.g., a non-crystalline plastic such as ABS resin). The terminals (**131b**, **132b**) are used to extend the wirings. The gate housing **130** and the main housing **135** have a pair of a module port **134** and a module connector **142**, which are detachably connected to each other to simultaneously establish both of supplying the electric power from the gate housing **130** to the main housing **135**, and making a signal transmission therebetween. In addition, when the function unit **104** has the module connector **142**, it can be connected to the gate housing **130** in place of the main housing **135**. If necessary, the gate housing **130** and main housing **135** may be integrally formed.

Circuit components of the base unit **103** are designed to transmit the electric power and the information signal to the function unit **104**. For example, the base unit **103** is provided with an AC/AC converter **160**, DC power section **161**, transceiver section **162**, E/O converter **163**, O/E converter **165**, and a function portion **167**.

The AC/AC converter **160** converts commercial AC voltage to a lower AC voltage having an increased frequency, and applies the lower AC voltage to a coil

172. The DC power section **161** generates an operating voltage of the internal circuit components from a stable DC voltage obtained by rectifying and smoothing the lower AC voltage. The transceiver section **162** transmits and receives the information signal for enabling the mutual communication through the information line **Ls**. The E/O converter **163** converts the information signal received from the information line **Ls** to an optical signal, and outputs the optical signal though a light emitting device (LED) **164**. On the other hand, the O/E converter **165** receives the optical signal provided from the outside, e.g., the function unit **104** by a light receiving device (PD) **166**, and converting the received optical signal into the information signal to transmit it to the transceiver section **162**. The base unit **103** shown in FIG. 23 has an outlet tap as the function portion **167**. Alternatively, a sensor device or a controller may be formed as the function portion **167**.

The module port **134** formed at a front surface of the gate housing **130** is composed of an electric power port **134a** for supplying the electric power and an information signal port **134b** for accessing the information line **Ls**, as shown in FIG. 25B. Arrangement and shapes of the electric power port **134a** and the information signal port **134b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25B, each of the electric power port **134a** and the information signal port **134b** is configured in a substantially rectangular shape such that they are arranged in parallel to each other.

On the other hand, the module connector **142** formed at a rear surface of the main housing **135** is composed of an electric power connector **142a** and an information signal connector **142b**, as shown in FIG. 25A. Arrangement and shapes of the electric power connector **142a** and the information signal connector **142b** are standardized (normalized or stylized) in the wiring system of this embodiment. For example, as shown in FIG. 25A, each of the electric power connector **142a** and the information signal connector **142b** is configured

in a substantially rectangular shape such that they are arranged in parallel to each other.

In this embodiment, the module port **134** has a guide portion **133** such as a ring-like wall or a ring-like groove extending around the electric power port **134a** and the information signal port **134b**. This guide portion **133** is formed to be engageable to an engaging portion **145** such as a ring-like wall of the main housing **135**, which is formed around the electric power connector **142a** and the information signal connector **142b**. By simply engaging the engaging portion **145** to the guide portion **133**, the electric power connector **142a** and the information signal connector **142b** can be simultaneously connected to the electric power port **134a** and the information signal port **134b**. Therefore, there advantages that the main housing **135** can be easily attached to or removed from the gate housing **130** with improved connection reliability. A module connector having the same structure described above may be formed in the function unit **104**. The module port **134** and the module connector **142** may be formed by female and male connectors.

In the present embodiment, the gate housing **130** is directly fixed to the switch box **102**. If necessary, the gate housing **130** may be fixed to the switch box **102** through an attachment member **175**, as shown in FIG. 26. For example, this attachment member **175** is formed by a rectangular frame **176** with a window hole **177**. In this case, the gate housing **130** is mounted on the attachment member **175** by fitting a front portion of the gate housing into the window hole **177**, and then engaging hooks (not shown) formed at both sides of the gate housing to the rectangular frame **176**. The attachment member **175** mounting the gate housing **130** thereon is fixed to the switch box **102** by use of fixing screws. Alternatively, the gate housing **130** may be directly fixed to the wall surface by use of exclusive clamps without using the switch box **102**.

The function unit **104** is designed to provide a desired function by using the electric power supplied to the function unit **104** through the base unit **103** or

using the mutual communication of the information signal between the function unit **104** and the information line **Ls** through the base unit **103**. For example, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a high position near the ceiling, it preferably has a receptacle function of receiving a plug with hook of a lighting apparatus, security function such as a motion sensor, temperature sensor, and monitoring camera, or a sound function such as speaker. In addition, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a middle height, at which the function unit **104** can be easily operated by the user, it preferably has a switch function of turning on/off the lighting apparatus, control function for electric appliances such as air-conditioning equipments, a display function such as liquid crystal display or a timer function. The intercom device of the present invention is preferably connected as one of the function units **104** to the base unit **103** at the middle height. Moreover, when the function unit **104** is connected to the base unit **103** mounted in the wall surface at a low position near the floor, it preferably has a receptacle function for receiving a plug of an electric appliance such as electric vacuum cleaner, the sound function such as speaker, or a footlight function.

Specifically, as shown in FIG. 27, when a function section **181** of the function unit **104** is formed by a switch, operation data obtained by operating the switch is transmitted to a processing section **188** such as CPU through an I/O interface **189**. Then the processed data is sent to, for example, an infrared remote controller (not shown) through a transceiver section **187**, so that an electric appliance to be controlled is turned on/off by receiving a remote control signal emitted from the infrared remote controller. In addition, when the function section **181** is formed by a sensor, data detected by the sensor is transmitted as the information signal to the information line **Ls**, and then informed to the user by a required communicator.

The coil **172** wound around a core **170** of the base unit **103** shown in FIG.

23 is used as a power transmission means for supplying electric power to the function unit **104** in a noncontact manner. That is, the coil **172** of the base unit **103** provides an electromagnetic coupling portion that works as a first side of a transformer. On the other hand, as shown in FIG. 27, the function unit **104** has an electromagnetic coupling portion comprised of a coil **182** wound around a core **180**, which works as a second side of the transformer. Therefore, by forming electromagnetic coupling between the base unit **103** and the function unit **104**, a low AC voltage can be induced in the coil **182** of the function unit **104** to supply of electric power from the base unit **103** to the function unit **104**. Similarly, an electromagnetic coupling portion **X2** is formed at the opposite side of the electromagnetic coupling portion **X1**, and used to make a connection between the function units. In this embodiment, since the low AC voltage having the higher frequency than the commercial AC voltage is obtained by the AC/AC converter **160**, there is an advantage that the electromagnetic coupling portions used as the transformer can be downsized.

In addition, the light emitting device (LED) **164** of the base unit **103** shown in FIG. 23 is used to transmit an optical signal as the information signal to the function unit **104** in a noncontact manner. On the other hand, as shown in FIG. 27, a light receiving device (PD) **186** is disposed in the function unit 10 such that the light emitting device **164** is in a face-to-face relation with the light receiving device **186** when the function unit **104** is connected to the base unit **103**. Similarly, to transmit the optical signal as the information signal from the function unit **104** to the base unit **103**, the function unit **104** has a light emitting device (LED) **184**, which is disposed in the face-to-face relation with the light receiving element (PD) **166** when the function unit **104** is connected to the base unit **103**. Thus, each of the base unit **103** and the function unit **104** has the pair of the E/O converter (**163**, **183**) and the O/E converter (**165**, **185**) as an optical coupling portion to enable the mutual communication of the information signal therebetween in a noncontact manner.

As shown in FIG. 23, the electromagnetic coupling portion **X** used to supply the electric power and the optical coupling portion **Y** used for the mutual communication of the information signal are disposed at a side surface of the base unit **103** to be spaced from each other by a constant distance. Arrangements and shapes of the electromagnetic coupling portion **X** and the optical coupling portion **Y** are standardized (or stylized) to make the base unit **103** shareable among the function units **104**. In addition, the pair of the electromagnetic coupling portion **X** and the optical coupling portion **Y** may be formed at each of both sides of the function unit **104**, as shown in FIG. 27. That is, the optical coupling portion **Y1** formed at one side (e.g., left side) of the function unit **104** is composed of the light receiving device **186** located at the upper side and the light emitting device **184** located at the lower side, and the optical coupling portion **Y2** formed at the opposite side (e.g., right side) of the function unit **104** is composed of a light emitting device **194** located at the upper side and a light receiving device **196** located at the lower side. In this case, since the base unit **103** is connected to the one side of the function unit **104**, and another function unit or the intercom device of the present invention can be connected to the other side of the function unit **104**, there is an advantage of further improving function expandability of the wiring system.

The intercom device **101** of this embodiment can be connected to the base unit **103**, as shown in FIG. 28. In addition, internal components of the intercom device **101** of this embodiment are shown in FIG. 29. As understood from the comparison with the function unit **104** of FIG. 27, the intercom device **101** of FIG. 29 is substantially equivalent to a function unit having the components of the intercom device shown in FIG. 2 as the function section **181**.

Since this intercom device **101** has the stylized connectors as well as the function unit **104** and base unit **103** described above, it can be used in a desired combination with the base unit **103** and the function unit **104**. For example, as shown by the arrow ① in FIG. 30, the intercom device **101** can be

connected to the function unit **104** having the timer function, which is connected to the base unit **103**. Alternatively, as shown by the arrow ② in FIG 30, the intercom device **101** may be connected at its one side to the base unit **103**. At this time, the function unit **104** having the timer function can be connected to the other side of the intercom device **101**. Thus, there is a remarkable advantage that a layout of the function units **104** having various functions and the intercom device **101** can be changed with a high degree of freedom according to the user's needs.

In addition, when the intercom device **101** is operated in conjunction with at least one functions unit **104**, the wiring system of this embodiment can provide a higher order function in addition to the communication function. For example, when the intercom device **101** is connected to one of the base units **103**, and the function unit **104** having a sensor function (e.g., motion sensor or fire sensor) is connected to another one of the base units **103**, which is disposed at a proper location, the intercom device **101** can output an alarm sound by receiving alarm signal from the function unit. Therefore, the intercom device **101** of this case has an alarm generating function for disaster or crime prevention as well as the communication function. In addition, since it is not needed to individually prepare a speaker for generating the alarm sound, there is a further advantage of achieving an improvement in cost performance of the wiring system as a whole.

Next, means for mechanically coupling between the base unit **103** and the function unit **104** (or the intercom device **101**) is explained.

For example, the base unit **103** shown in FIG. 31 has a cosmetic cover **112**, which is detachably attached to a front surface of the gate housing **135**, and horizontal guide rails **114** formed at upper and lower portions of the gate housing. The numeral **115** designates a stopper wall formed in a substantially center position in the longitudinal direction of the guide rail **114**. On the other hand, as shown in FIGS. 32A to 32C, the function unit **104** (or the intercom

device **101**) has concave portions **128** for accommodating joining members **150** therein, which are formed at both of its upper and lower sides, horizontal guide rails **124** extending in the concave portions **128**, and cover members **126** pivotally supported at upper and lower ends of the function unit **104**. In addition, as shown in FIG. 32D, the joining member **150** has a groove **152**, in which the horizontal guide rail (**114**, **124**) can be slidably fitted.

In this case, a cosmetic cover **112** is firstly removed from the gate housing **135** of the base unit **103**. Then, the electromagnetic coupling portion **X** and the optical coupling portion **Y** of the base unit **103** are connected to the electromagnetic coupling portion **X1** and the optical coupling portion **Y1** of the function unit **104** (or the intercom device **101**). Next, the cover member **126** is opened, and the joining member **150** is slid along the guide rail **124**, as shown by an arrow in FIG. 33. The slide movement of the joining member **150** is performed until the joining member **150** contacts the stopper wall **115**. As a result, the joining member **150** is engaged to the guide rail **114** of the base unit **103** over about half length of the joining member **150**. On the other hand, the remaining portion of the joining member **150** is still engaged to the guide rail **124** of the function unit **104** (or the intercom device **101**). Thus, by engaging the joining member **150** to both of the base unit **103** and the function unit **104** the base unit **103** can be mechanically coupled with the function unit **104** (or the intercom device **101**). Finally, the cosmetic cover **112** is attached to the base unit **103**, and the cover member **126** is closed.

According to this coupling means, since the joining member **150** is held between the cosmetic cover **112** and the gate housing **135**, it is possible to prevent accidental falling of the joining member **150**, and obtain the stable mechanical connection therebetween without spoiling the beauty of them. In addition, since the joining member **150** is always accommodated in the concave portion **128** of the function unit **104** (or the intercom device **101**), there is no worry about loss of the joining member **150**. This coupling means is also usual

in the case of mechanically coupling between the function units **104**, or between the function unit **104** and the intercom device **101**.

As an information-signal transmitting method available in the wiring system, a baseband transmission or a broadband transmission can be used. In addition, the protocol is not limited to a specific one. For example, sound and visual information signals may be transmitted and received according to JT-H232 packet to make the mutual communication between the intercom devices. In a control system, it is also preferred to use a routing protocol for a broadcast or a unicast that controlling can be performed at a control ratio of 1:1 or 1:N according to operation data. Alternatively, the protocol used between the base units may be different from the protocol used in the function unit or the intercom device connected to the base unit. In this case, a protocol conversion is preferably performed by the base unit.

In addition, the wiring system of the above embodiment is essential to use both of the information line and the power line installed in the building structure. As a modification of this embodiment, a power-line carrier communication may be used in the wiring system.

That is, the wiring system according to this modification is essential to install only the power line in the building structure. Therefore, the base unit is connected to only the power line. On the other hand, the base unit, the function unit and/or the intercom device need to have a transceiver unit for transmitting and receiving information signals in the power-line carrier communication manner. For example, when the transceiver unit is formed in the base unit, the electric power carrying the information signals thereon is separated to the electric power transmission and the information signal transmission by the base unit. Therefore, the same function unit and the intercom device used in the wiring system of the above embodiment are also available in this power-line carrier type wiring system.

Alternatively, the transceiver unit may be formed in the intercom device **101**, as shown in FIG. 34. In this case, the intercom device is detachably

connected to the base unit and/or the function unit by a power-line carrier type connector **Z**. In addition, the intercom device also comprises a PLC modem **210** for receiving the information signals carried by the power line, and transmitting information signals through the power line, a processing section **220** connected to the PLC modem, an I/O interface **230** between the processing section, and a function section **240** for providing the intercom function. Similar circuit components may be formed in the function unit and/or the base unit. As a modulation method for the power-line carrier communication, for example, it is possible to use a broad spectrum diffusion model, multi carrier model or an orthogonal Frequency division multiplexing (OFDM) model.

INDUSTRIAL APPLICABILITY

Thus, according to the present invention, the first microphone disposed in the front air chamber efficiently receives the voice output of the speaker, and the output of the first microphone is effectively used in the signal processing unit to remove the signal component corresponding to the voice output of the speaker from the output of the second microphone. Therefore, it is possible to improve the howling prevention effect. In addition, the intercom device can be downsized because a distance between the first microphone and the speaker becomes short. Furthermore, the intercom device is suitable for the wiring system with excellent function expandability, as described in the above embodiments.

Therefore, the present invention is expected to further increase the demand of the intercom device, and especially as one of the components of the wiring system of the next generation.

CLAIMS:

1. An intercom device comprising:
a housing;
a speaker accommodated in said housing such that a front air chamber is defined as a space enclosed by a front surface of said speaker and an inner surface of said housing, and a rear air chamber is defined as a space enclosed by a rear surface of said speaker and an inner surface of said housing;
a first microphone disposed such that its sound receiving surface faces the front air chamber;
a second microphone disposed such that its sound receiving surface faces outside of said housing; and
a signal processing unit configured to, when an audio output of said speaker is picked up by said second microphone, remove a signal component corresponding to the audio output of said speaker from an output signal of said second microphone by use of an output signal of said first microphone.

2. The intercom device according to claim 1, wherein the rear air chamber is the space hermetically-sealed by the rear surface of said speaker and the inner surface of said housing.

3. The intercom device according to claim 1, wherein said first microphone is arranged such that the sound receiving surface of said first microphone faces a diaphragm of said speaker.

4. The intercom device according to claim 1, wherein said second microphone is located adjacent to said speaker.

5. The intercom device according to claim 1, wherein said first microphone and said second microphone are mounted on a same circuit board.

6. The intercom device according to claim 5, wherein the circuit board is mounted on an outside surface of said housing such that the sound receiving surface of said first microphone faces the front air chamber and the sound receiving surface of said second microphone faces the outside of said housing.

7. The intercom device according to claim 1, further comprising an acoustic tube having an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

8. The intercom device according to claim 7, wherein said acoustic tube has a tube length of an odd multiple of one quarter of a wavelength determined in consideration of sonic velocity and an intended frequency where an increase in sound pressure level of the audio output of said speaker is needed.

9. The intercom device according to claim 1, further comprising a plurality of acoustic tubes having different lengths, each of which has an opening communicated with the rear air chamber at its one end, and a closed end at the other end.

10. The intercom device according to claim 7, wherein said acoustic tube has a resonant frequency between a frequency equivalent to a lowest resonant

frequency of said speaker that is supposed to be mounted on an infinite baffle and a lowest resonant frequency of said speaker mounted on the rear air chamber with the absence of said acoustic tube.

11. The intercom device according to claim 7, wherein said acoustic tube is formed along an inner surface of the rear air chamber.

12. The intercom device according to claim 7, further comprising a sound absorbing material disposed at the opening of said acoustic tube.

13. A wiring system using the intercom device as set forth in claim 1, comprising at least one transmission line installed in a building structure to transmit electric power and information signals, and a plurality of base units each adapted in use to be mounted in a wall surface of said building structure, and connected to said transmission line;

wherein the intercom device has a first connector detachably connectable to a second connector formed in each of said base units,

when said first connector is connected to said second connector of one of said base units, the intercom device functions such that the information signals provided through said transmission line are output as audio information from said speaker unit, and an output of said signal processing unit is sent out through said transmission line.

14. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units, said function unit having a third connector detachably

connectable to one of said second connector of said base unit, and an extended connector formed in the intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, or when said first connector of the intercom device is connected to said second connector of said base unit, and said third connector of said function unit is connected to said extended connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided through said transmission line, outputting information signals provided through said transmission line, and sending out information signals through said transmission line.

15. The wiring system as set forth in claim 14, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

16. The wiring system as set forth in claim 13, further comprising a function unit adapted in use to be connected to said transmission line through one of said base units,

said function unit has a third connector detachably connectable to said second connector of said base unit and an extended connector detachably connectable to said first connector of said intercom device,

when said third connector of said function unit is connected to said second connector of said base unit, and said extended connector of said function unit is connected to said first connector of the intercom device, said function unit provides at least one of functions for supplying electric power provided from said transmission line through said base unit, outputting information signals provided from said transmission line through said base unit, and sending out

information signals to said transmission line through said base unit, and the intercom device operates such that the information signals provided from said transmission line through said base unit and said function unit are output as audio information from said speaker unit, and the output of said signal processing unit is sent out to said transmission line through said base unit and said function unit.

17. The wiring system as set forth in claim 16, wherein said first connector and said third connector are stylized in shape and arrangement, and said second connector and said extended connector are stylized in shape and arrangement corresponding to the stylization of said first connector and said third connector.

18. The wiring system as set forth in claim 13, wherein said first and second connectors provide an electric power transmission between said base unit and the intercom device by means of electromagnetic coupling.

19. The wiring system as set forth in claim 13, wherein said first and second connectors provide a signal transmission between said base unit and the intercom device by means of optical coupling.

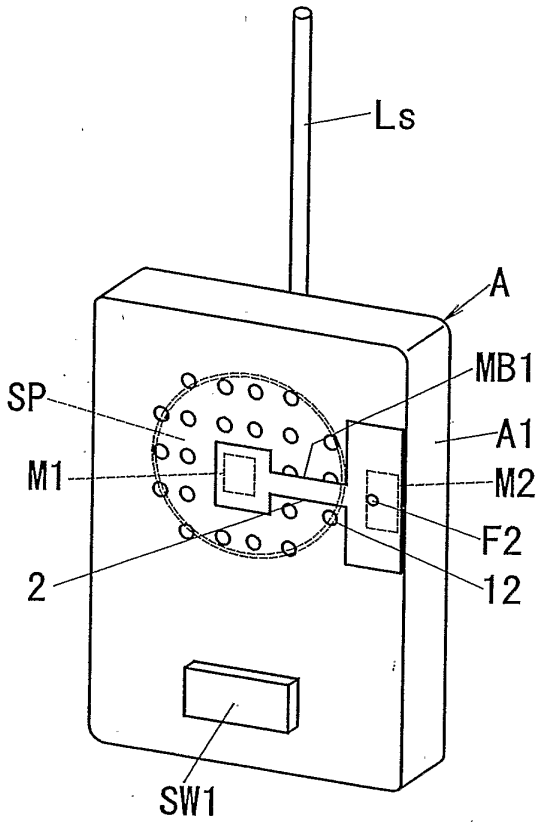


FIG. 1

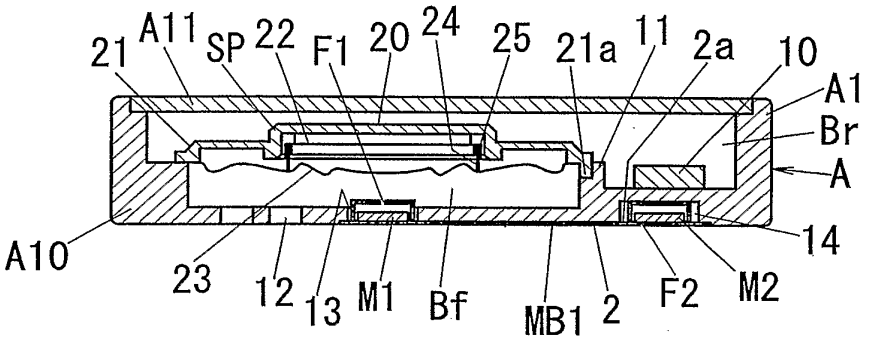

21
A11
SP
22
F1
20
24
25
21a
11
2a
10
A1
Br
A
14
A10
23
12
13
M1
Bf
MB1
2
F2
M2

FIG. 2

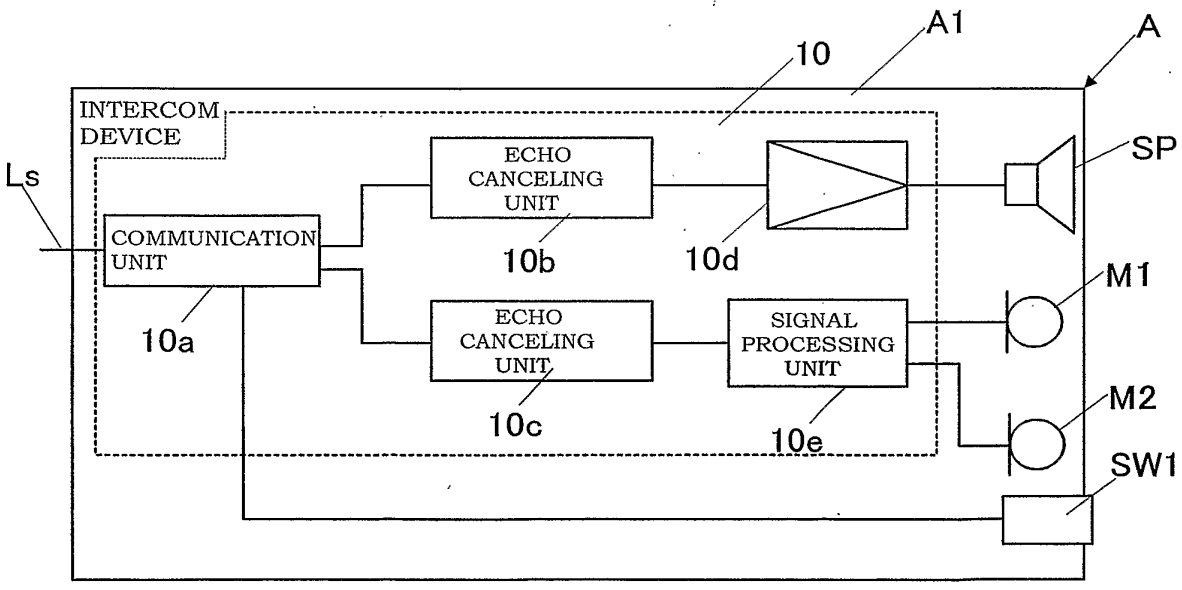

A1
10
A
INTERCOM DEVICE
Ls
ECHO CANCELING UNIT
SP
COMMUNICATION UNIT
10b
10d
M1
10a
ECHO CANCELING UNIT
SIGNAL PROCESSING UNIT
M2
10c
10e
SW1

FIG. 3

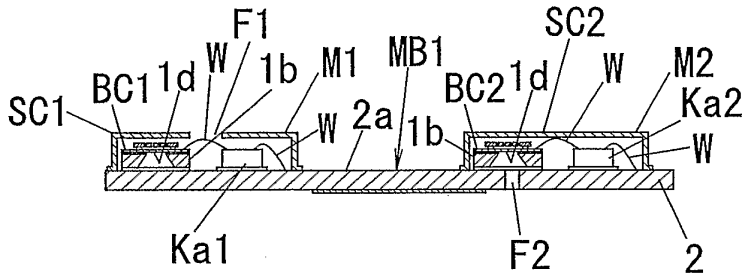

F1
W
1b
M1
MB1
SC2
W
M2
SC1
BC1
1d
W
2a
1b
BC2
1d
Ka2
W
Ka1
F2
2

FIG. 4

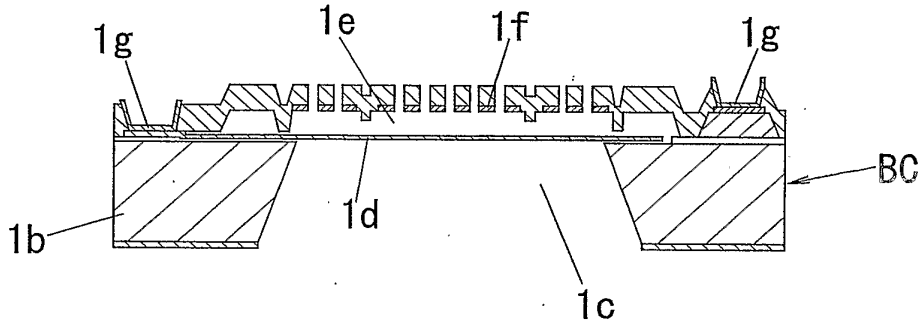

1g
1e
1f
1g
BC
1b
1d
1c

FIG. 5

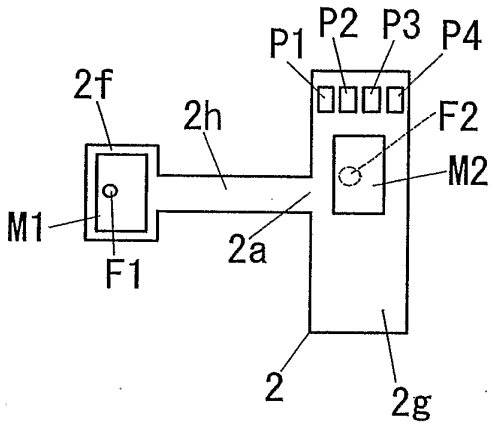


FIG. 6A

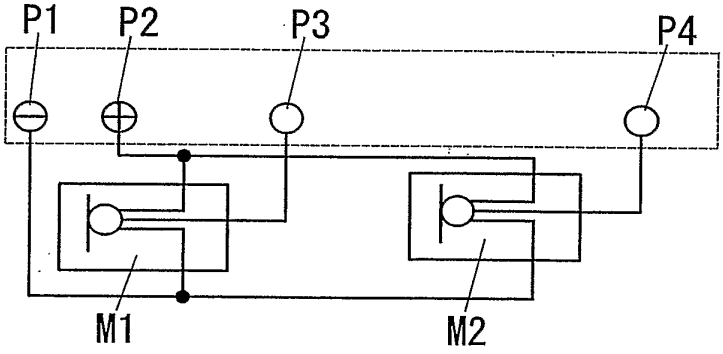


FIG. 6B

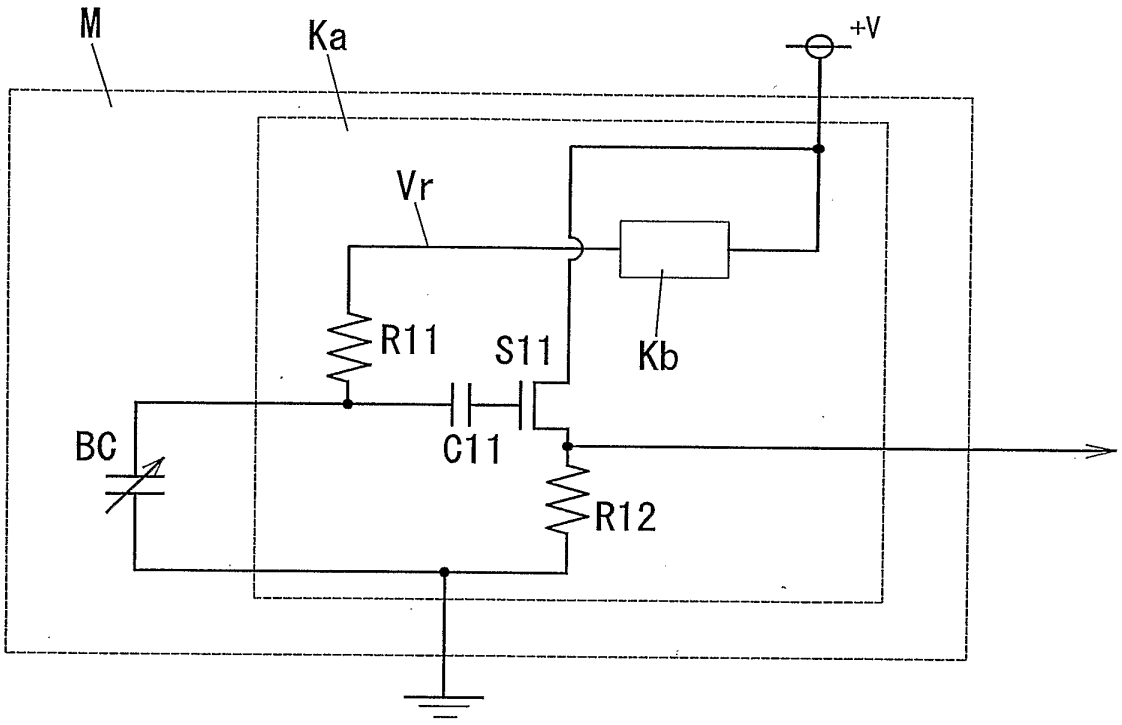


FIG. 7

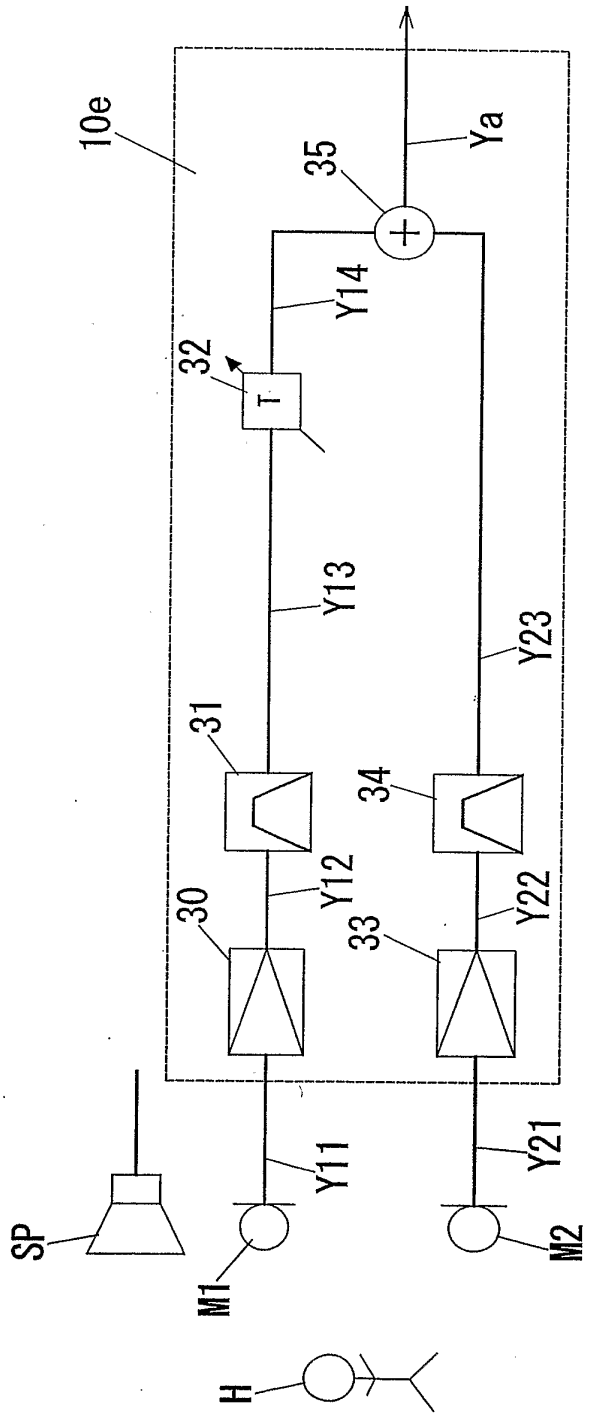


FIG. 8

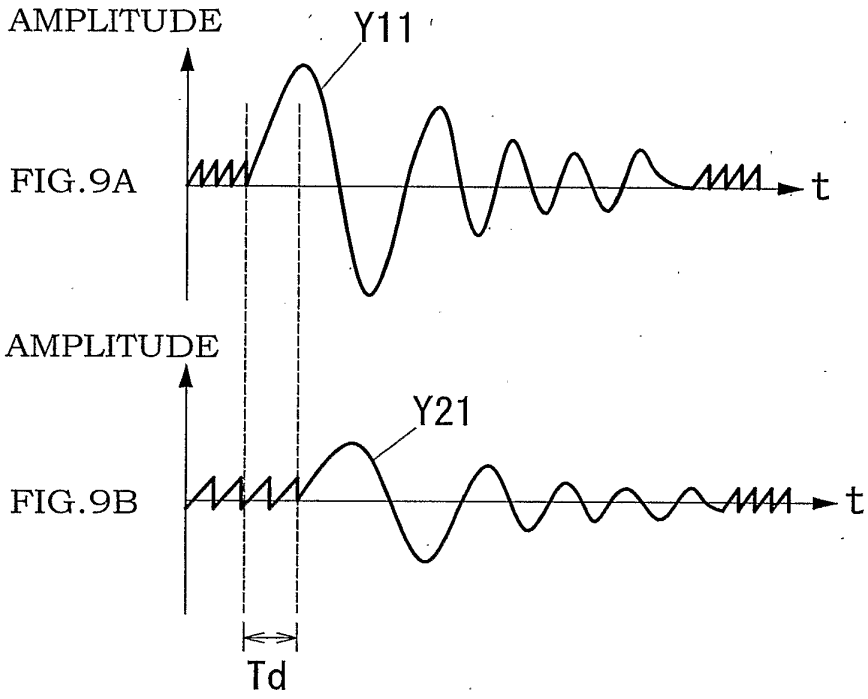


FIG.9A

FIG.9B

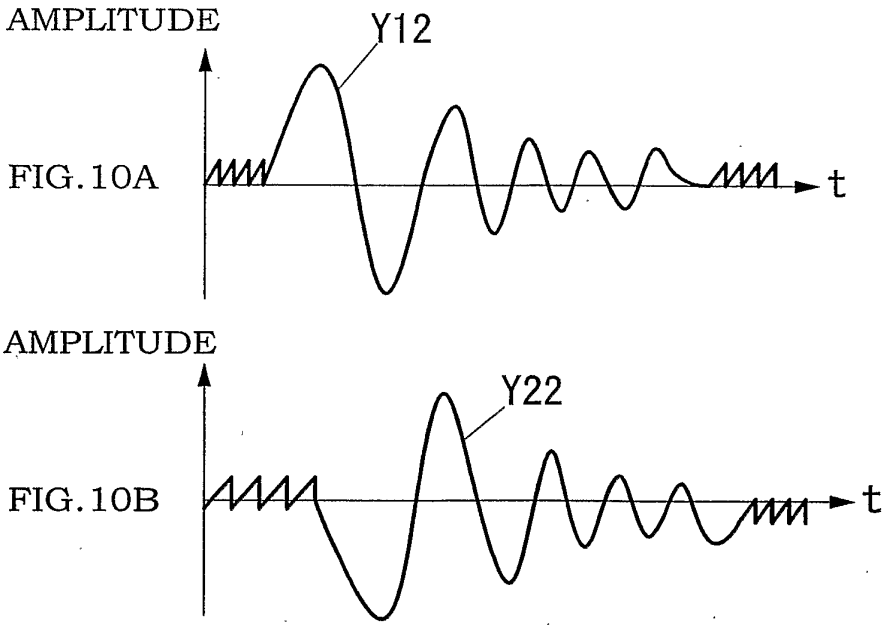


FIG.10A

FIG.10B

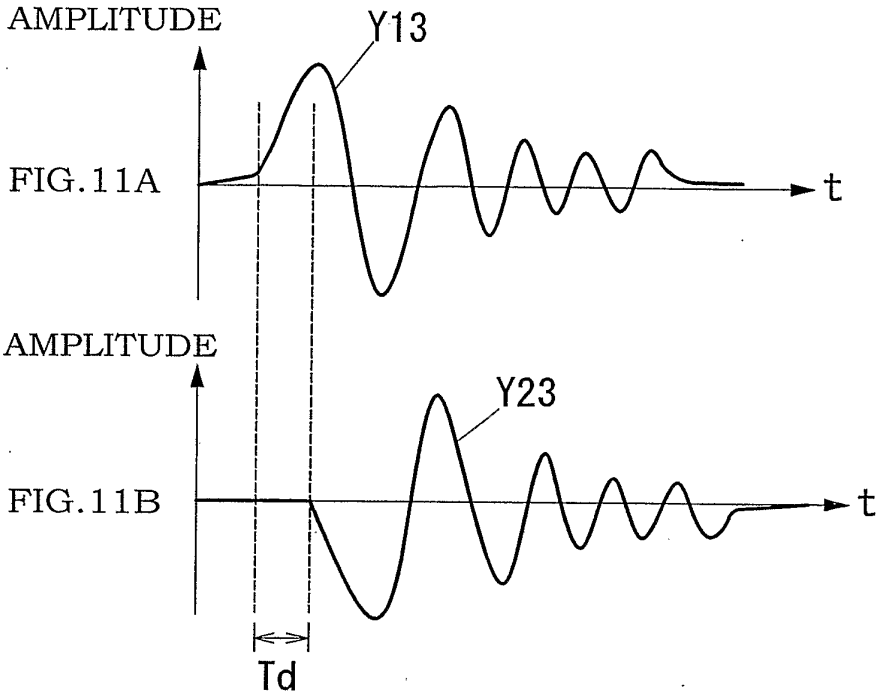


FIG.11A

FIG.11B

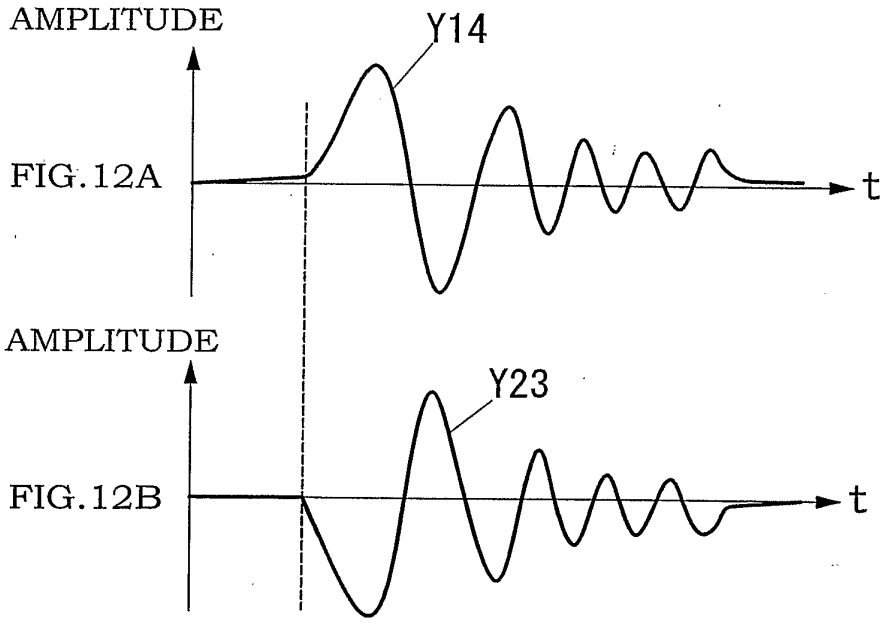


FIG.12A

FIG.12B

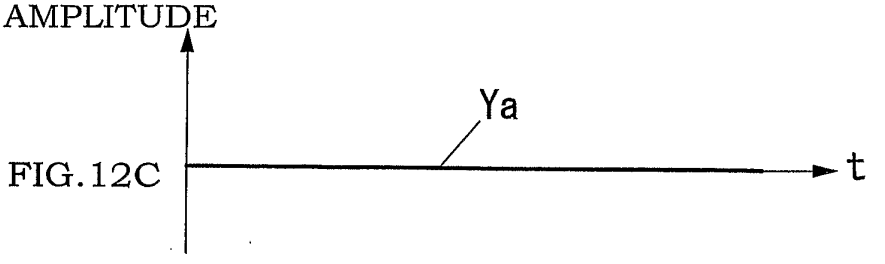
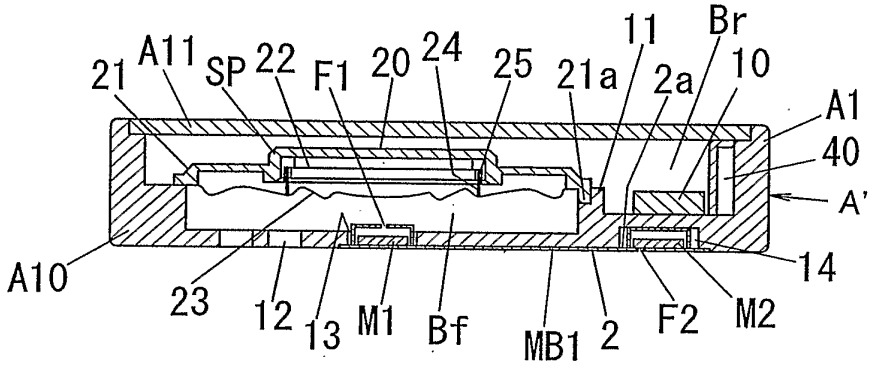


FIG.12C

FIG. 13



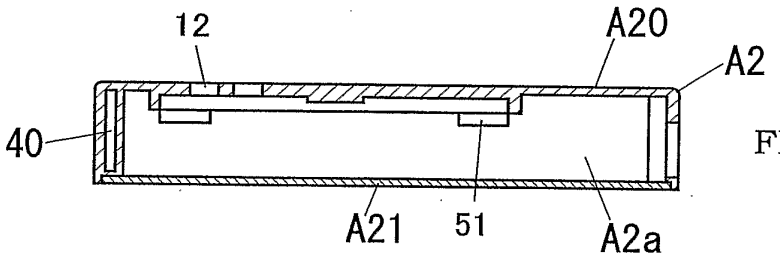
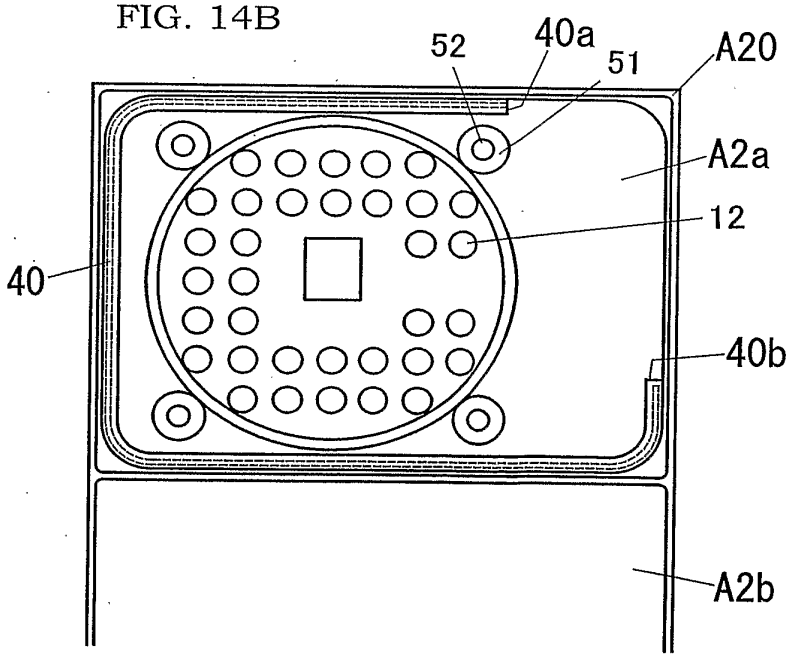


FIG. 14A

FIG. 14B



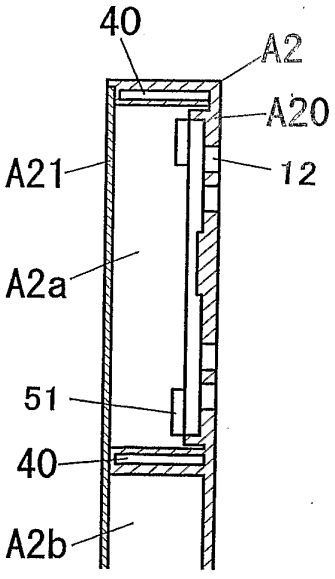


FIG. 14C

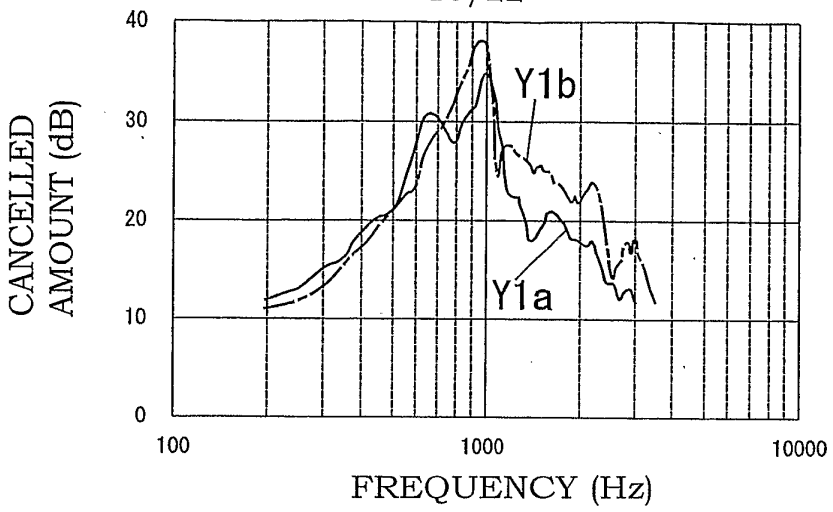


FIG. 15

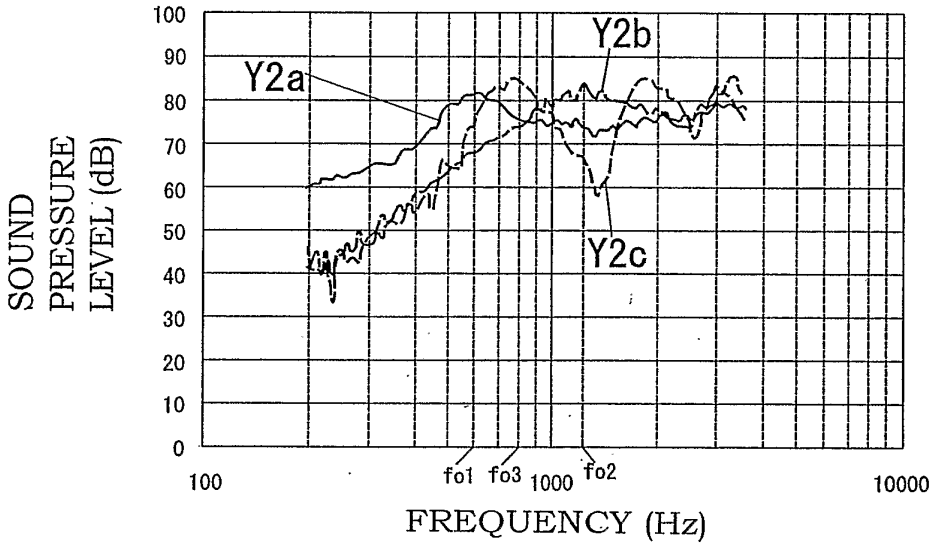


FIG. 16

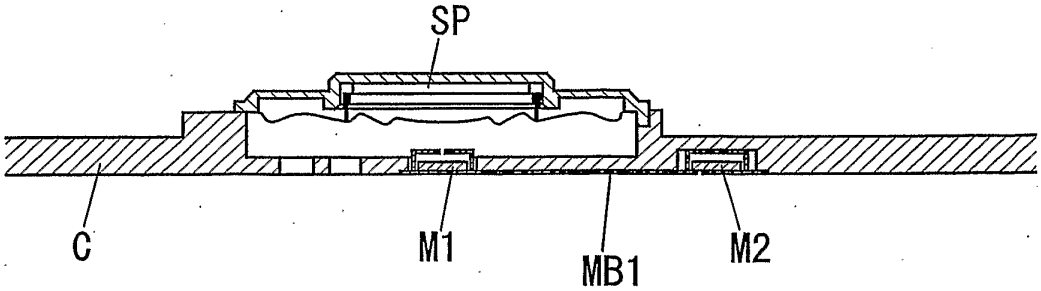


FIG. 17

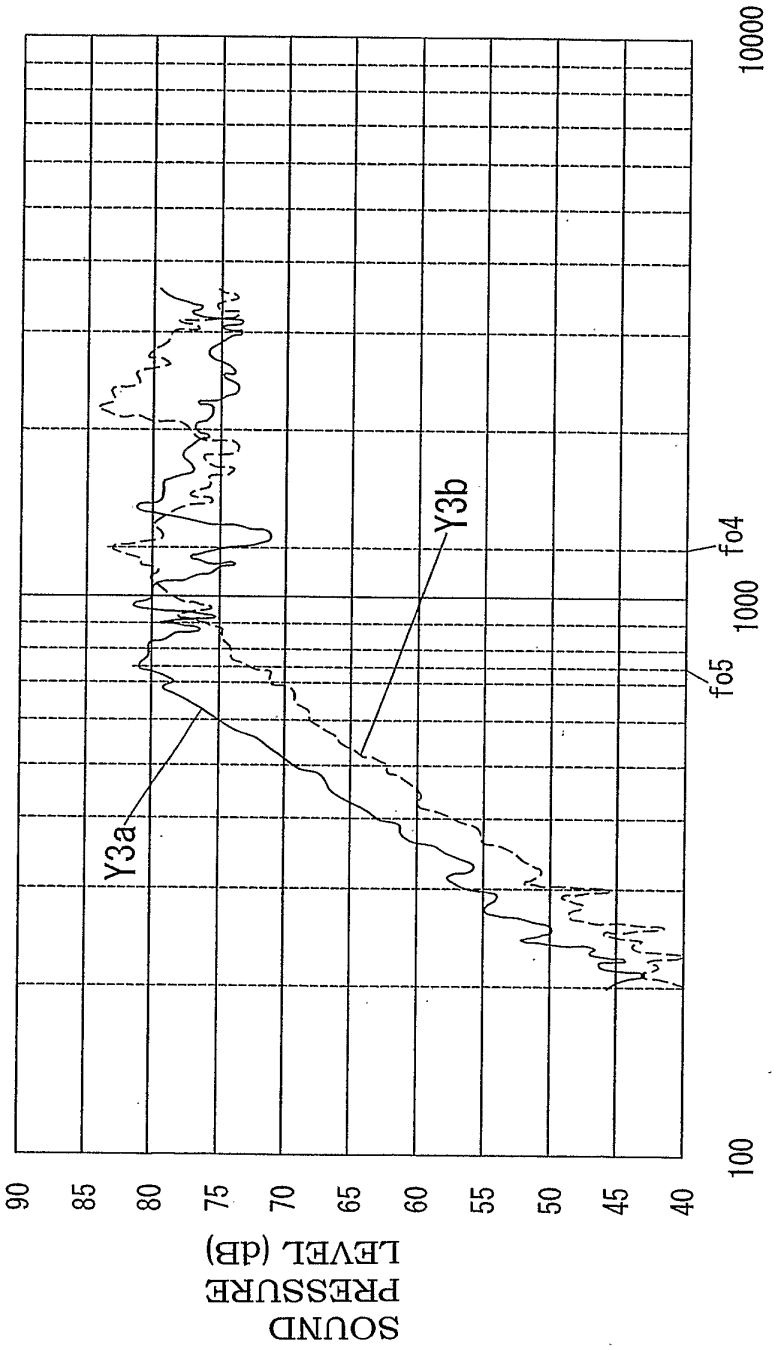

SOUND PRESSURE LEVEL (dB)
90
85
80
75
70
65
60
55
50
45
40
100
1000
10000
FREQUENCY (Hz)
fo4
fo5
Y3a
Y3b

FIG. 18

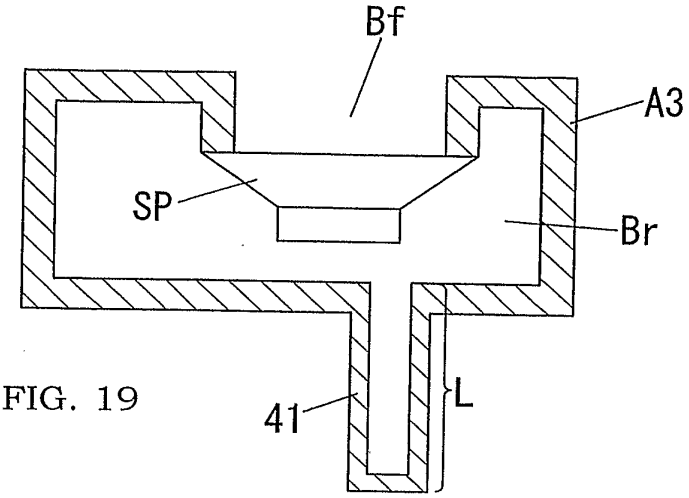


FIG. 19

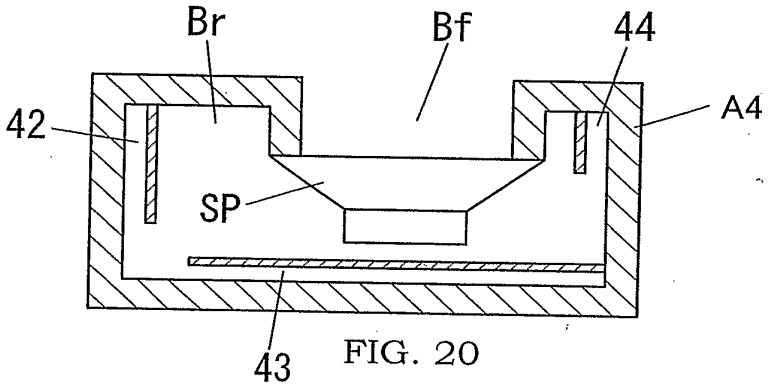


FIG. 20

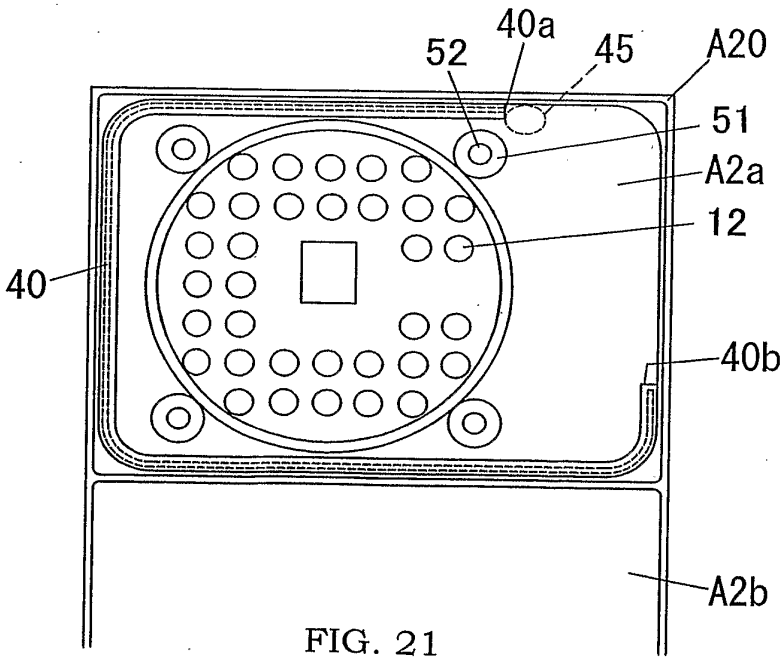


FIG. 21

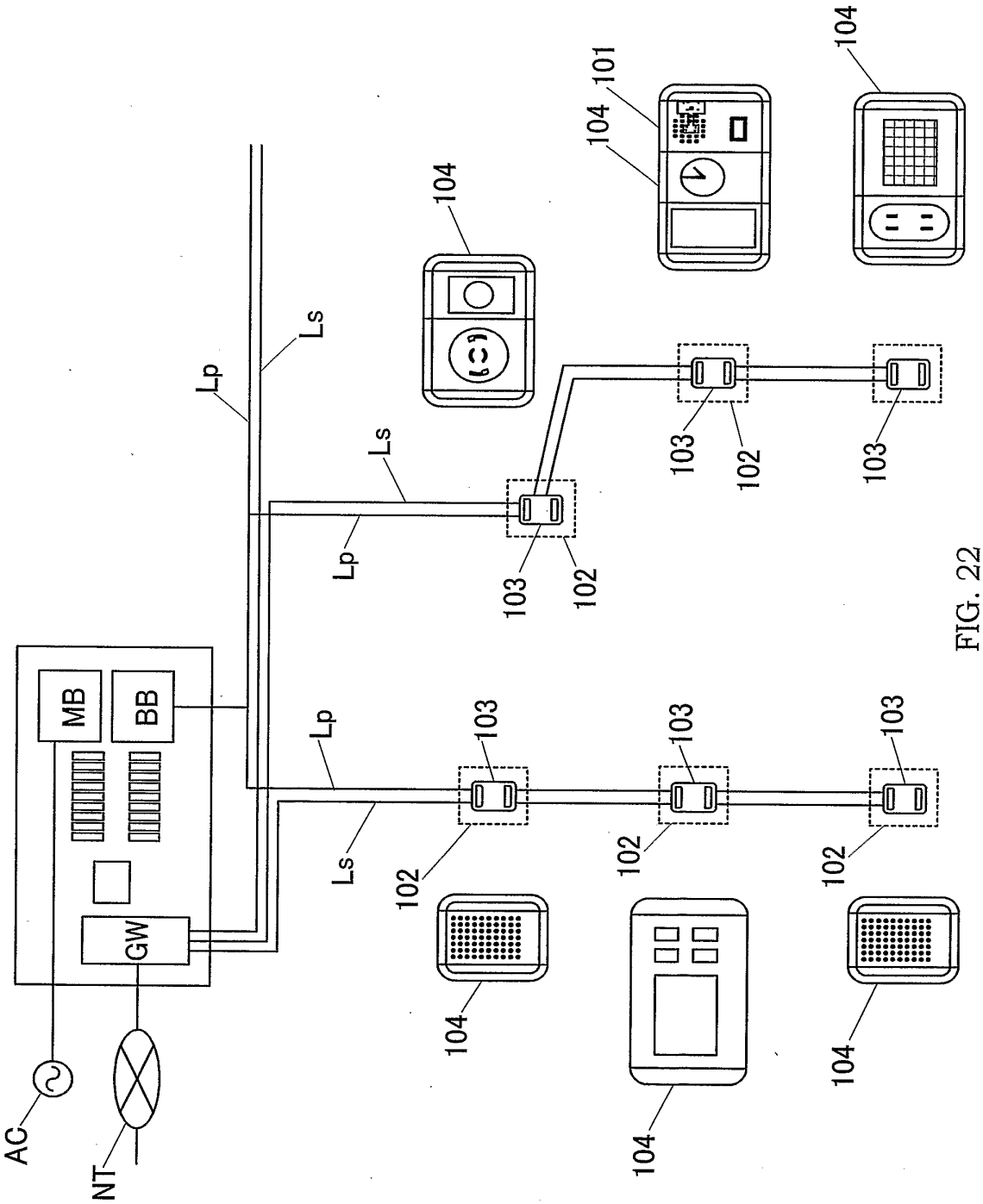


FIG. 22

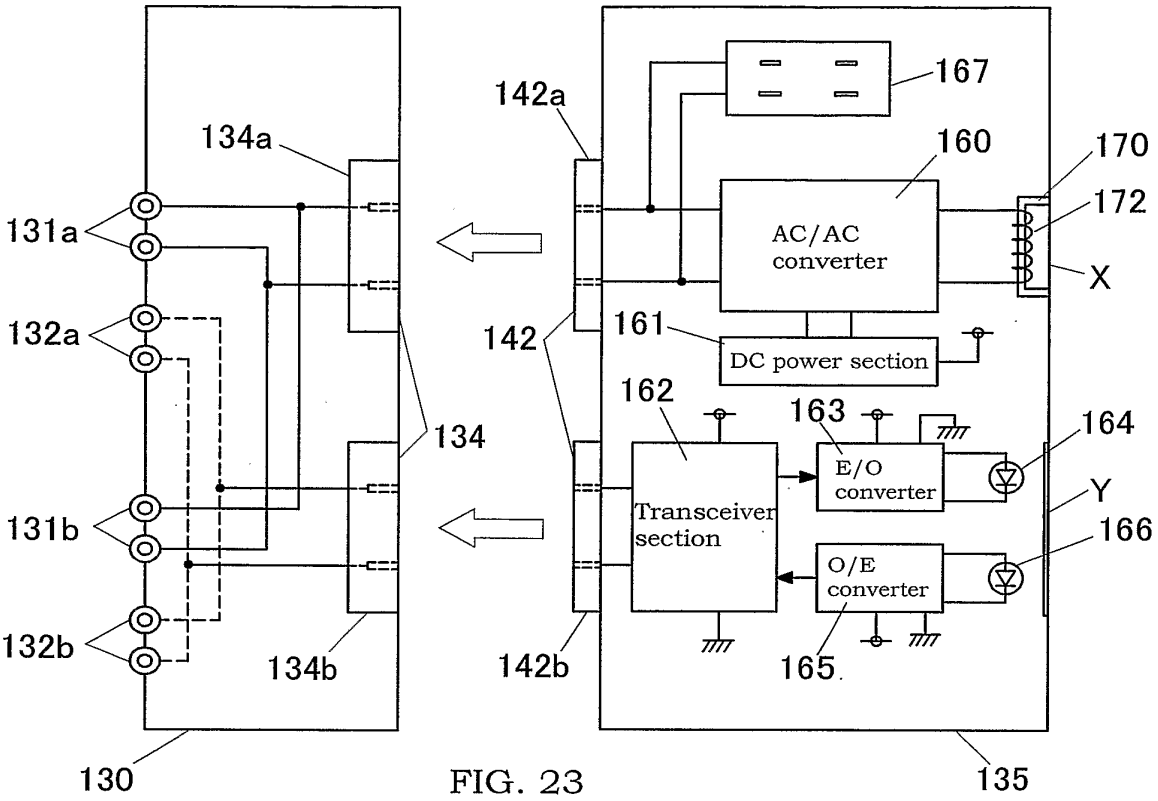

134a
131a
132a
131b
132b
134b
134
130
142a
167
160
170
172
AC/AC converter
X
142
161
DC power section
162
163
164
E/O converter
Y
Transceiver section
166
O/E converter
142b
165
135

FIG. 23

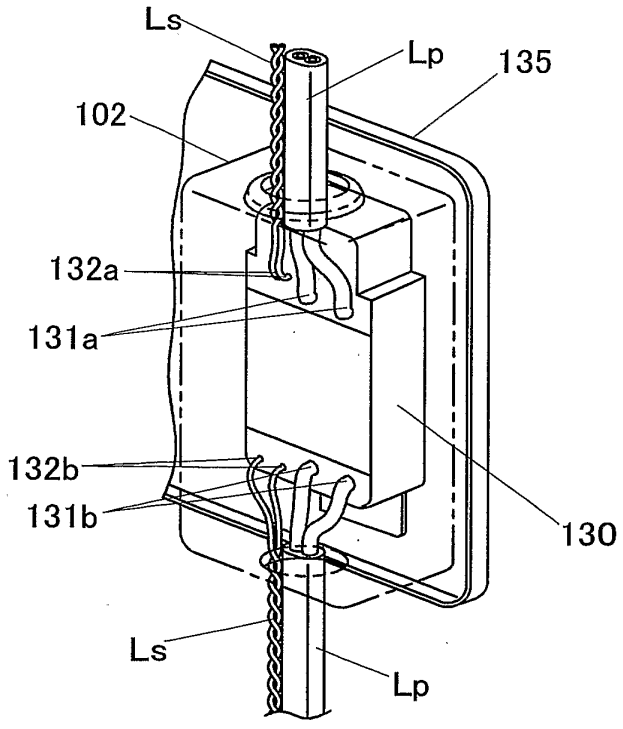

Ls
Lp
135
102
132a
131a
132b
131b
130
Ls
Lp

FIG. 24

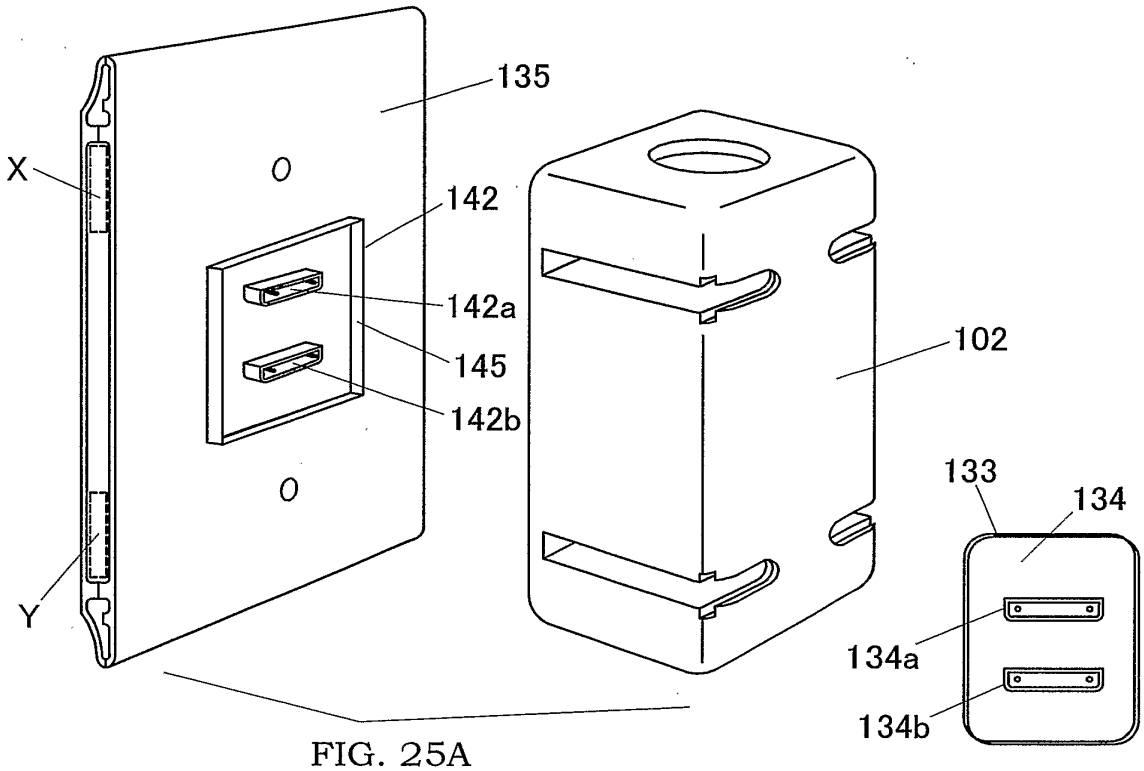


FIG. 25A

FIG. 25B

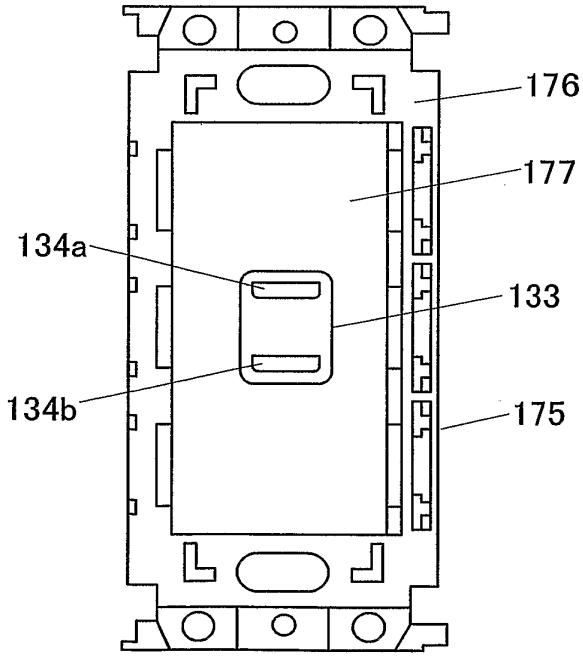


FIG. 26

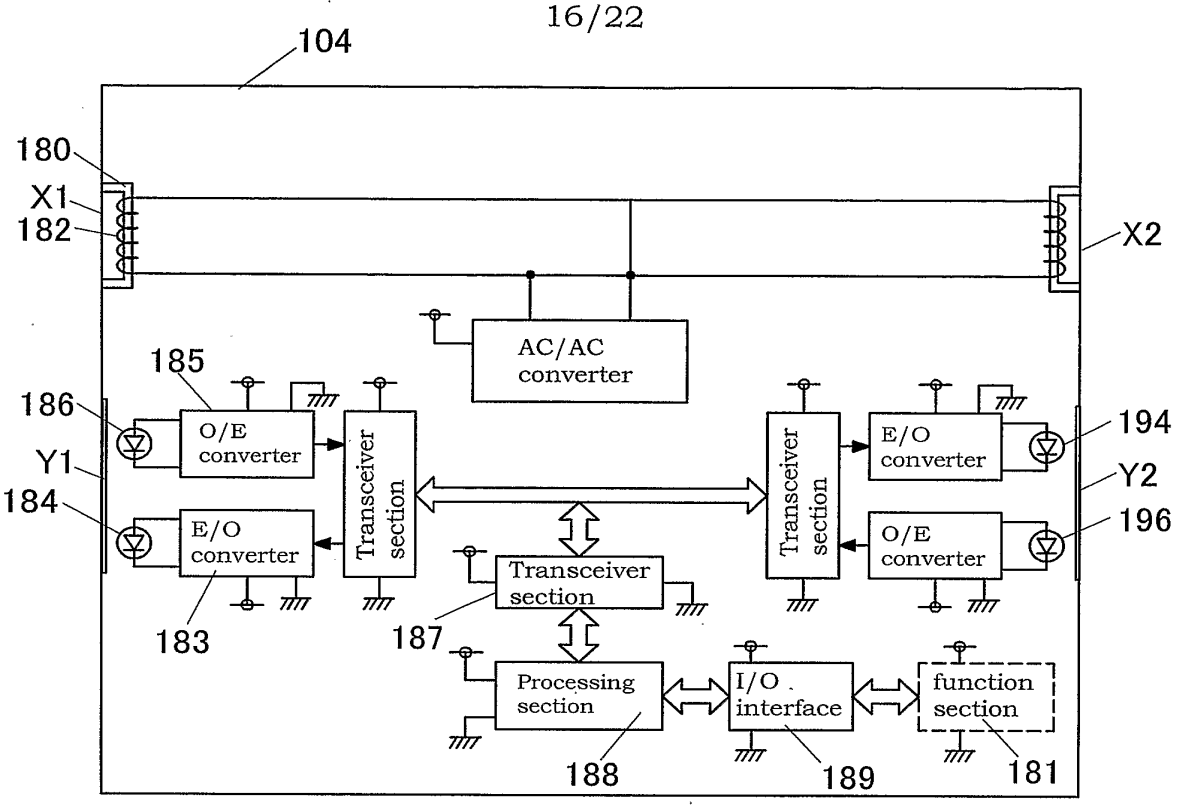

104
180
X1
182
X2
AC/AC converter
185
186
O/E converter
Transceiver section
Transceiver section
E/O converter
194
Y1
Y2
184
E/O converter
O/E converter
196
Transceiver section
183
187
Processing section
I/O interface
function section
188
189
181

FIG. 27

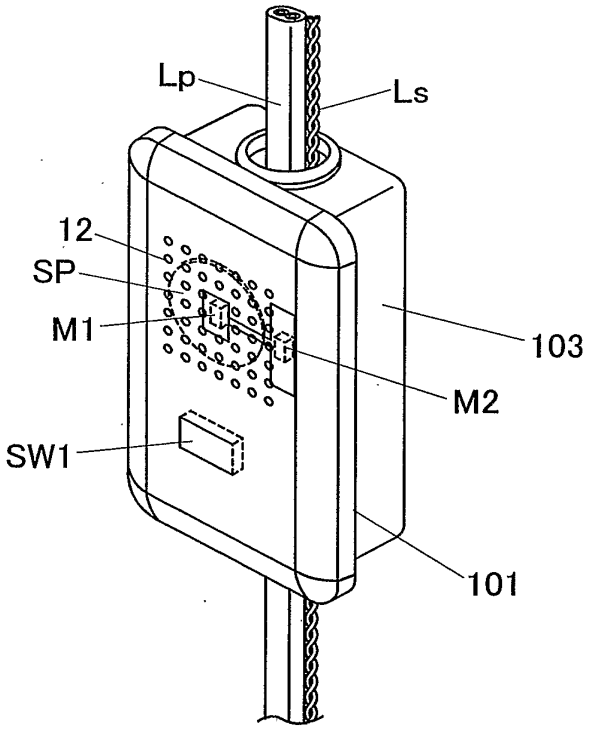

Lp
Ls
12
SP
M1
103
M2
SW1
101

FIG. 28

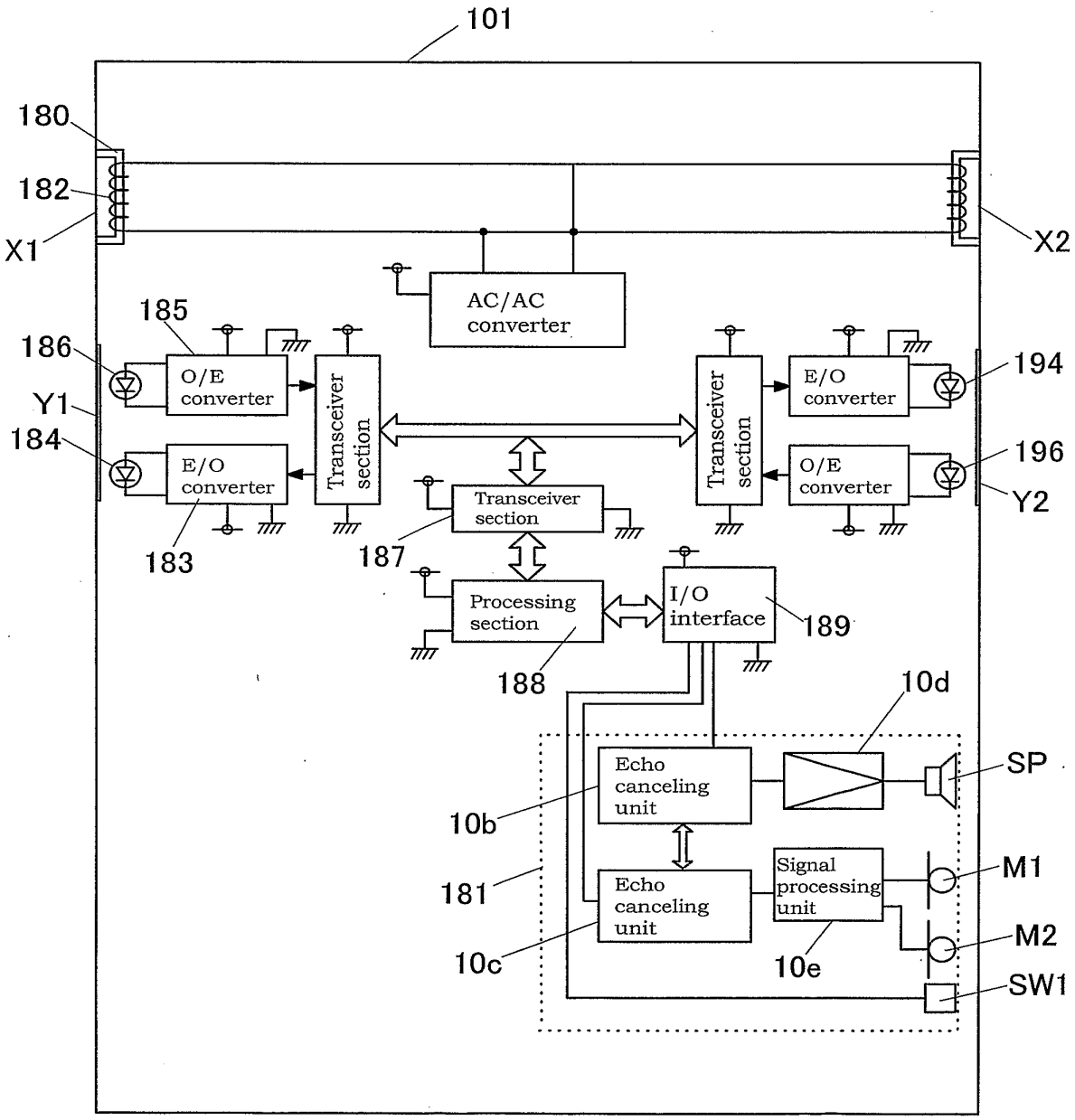

101
180
182
X1
X2
AC/AC converter
185
186
O/E converter
Transceiver section
Y1
184
E/O converter
183
Transceiver section
187
Transceiver section
E/O converter
194
O/E converter
196
Y2
Processing section
I/O interface
189
188
10d
Echo canceling unit
SP
10b
181
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1

FIG. 29

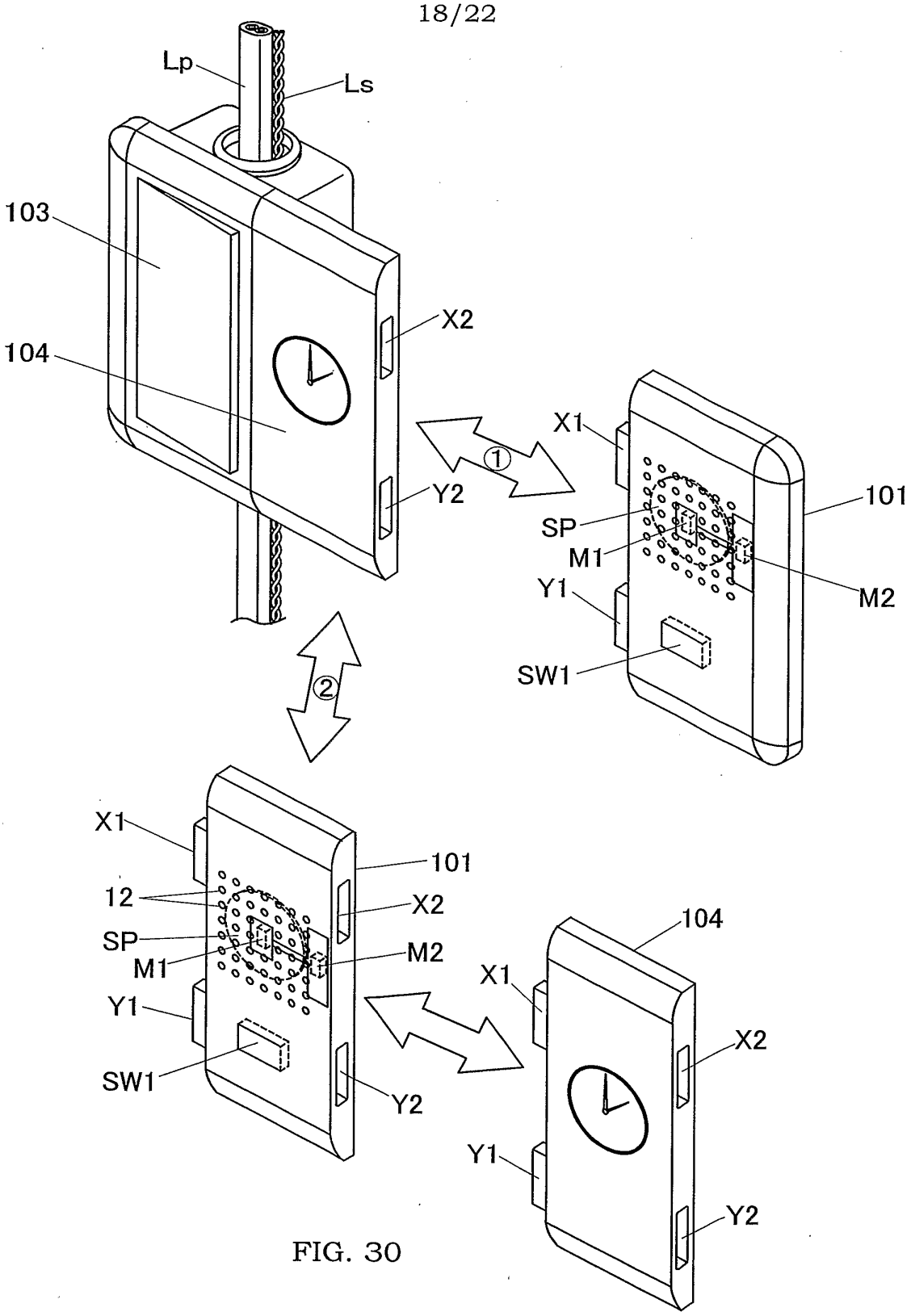

Lp
Ls
103
104
X2
Y2
①
X1
101
SP
M1
M2
Y1
SW1
②
X1
101
12
X2
SP
M1
M2
Y1
SW1
Y2
104
X1
X2
Y1
Y2

FIG. 30

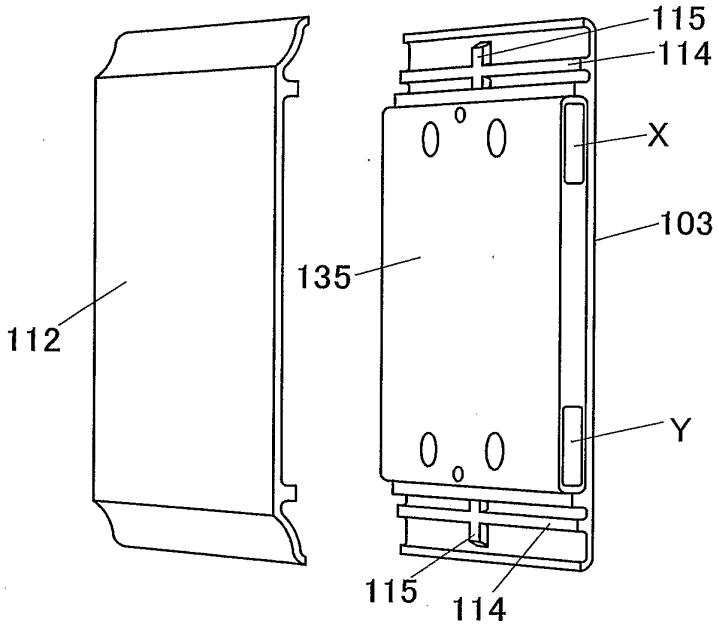


FIG. 31

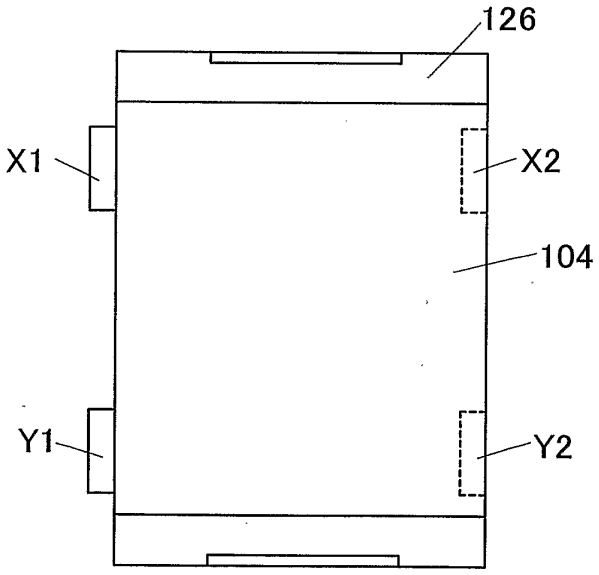


FIG. 32A

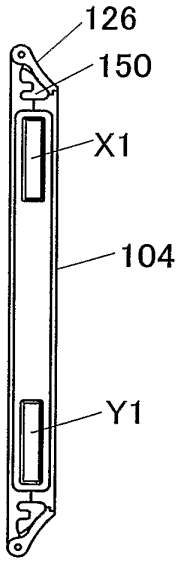


FIG. 32B

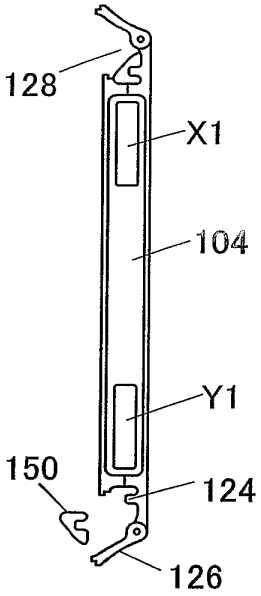


FIG. 32C

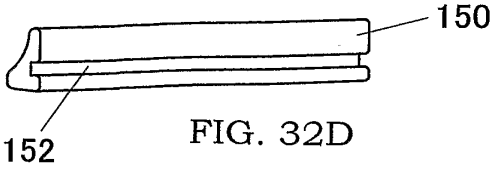


FIG. 32D

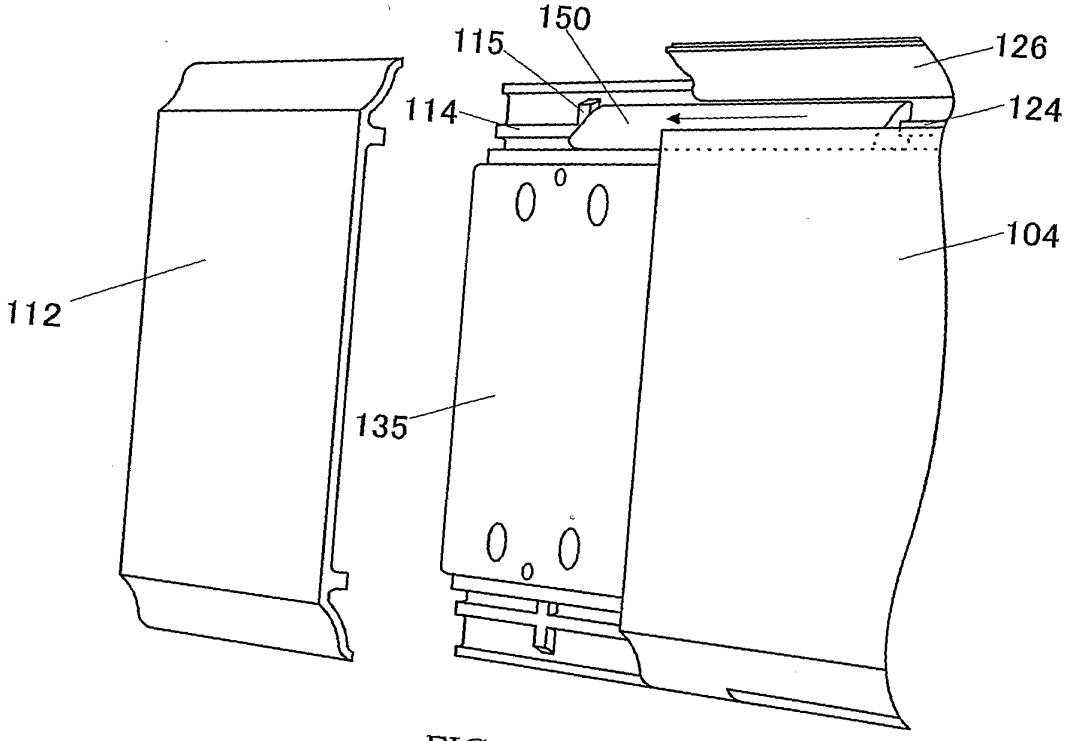


FIG. 33

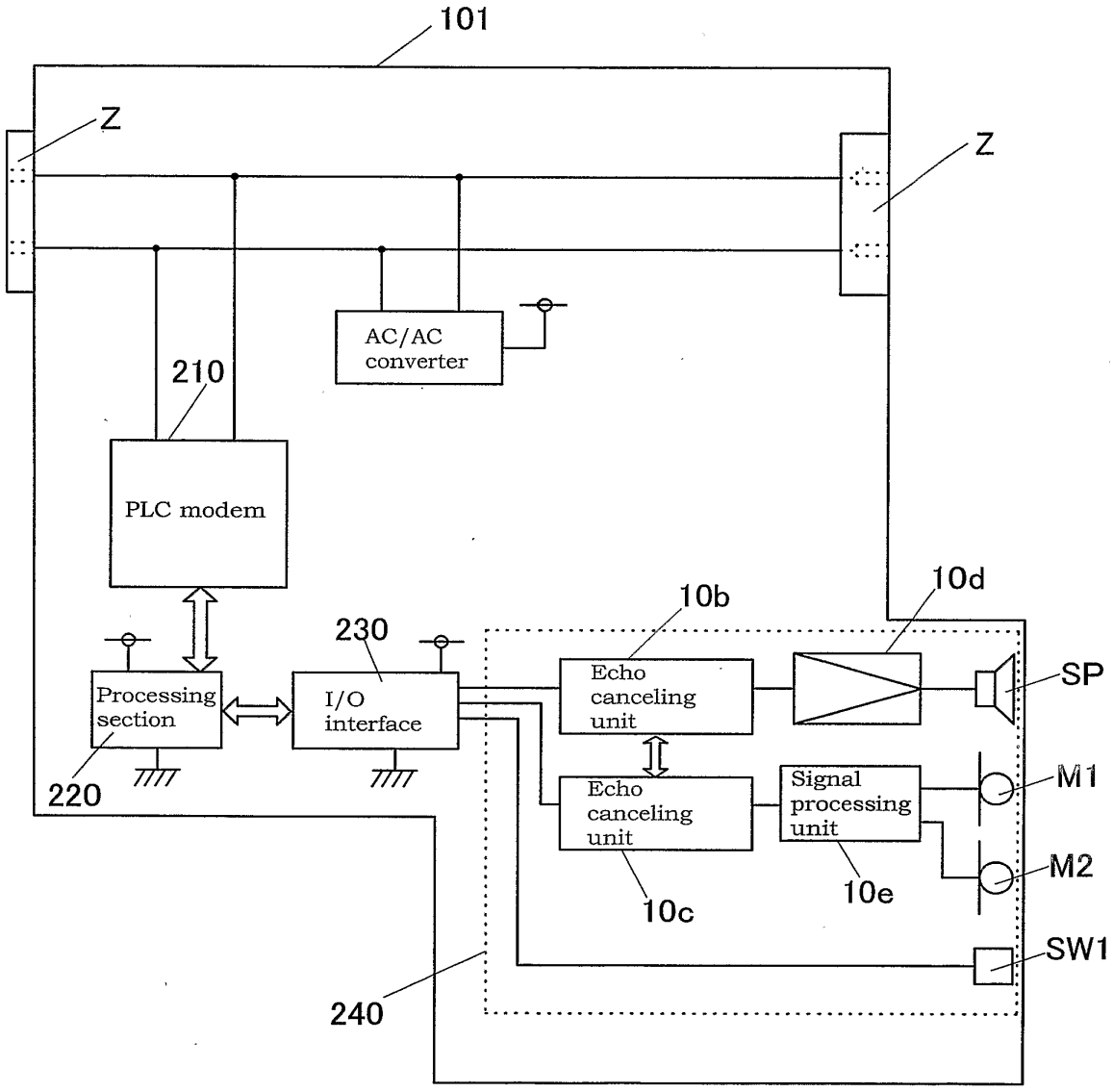

101
Z
Z
AC/AC converter
210
PLC modem
10b
10d
230
Processing section
I/O interface
Echo canceling unit
SP
220
Echo canceling unit
Signal processing unit
M1
M2
10c
10e
SW1
240

FIG. 34

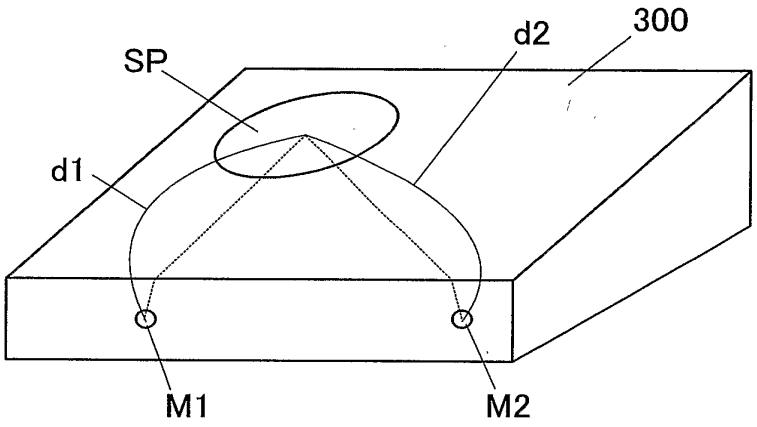


FIG. 35A

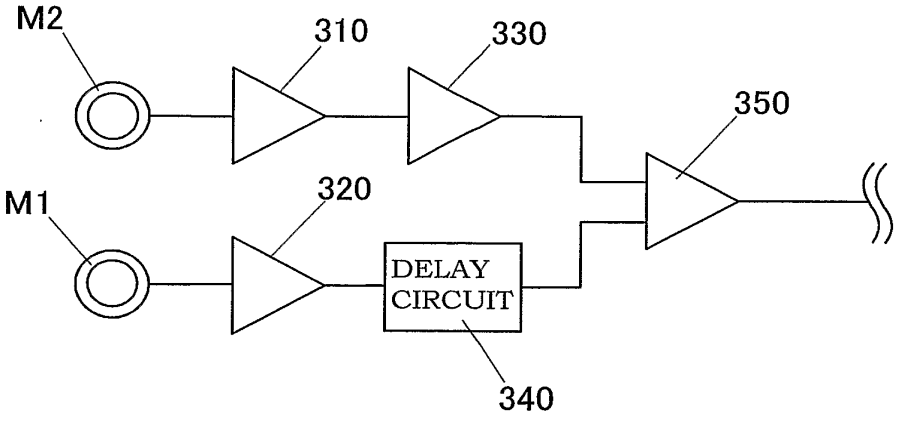


FIG. 35B

INTERNATIONAL SEARCH REPORT

International application No.
PCT/JP2007/069131

A. CLASSIFICATION OF SUBJECT MATTER

Int.Cl. H04R3/02(2006.01)i, H04M1/62(2006.01)i, H04R1/02(2006.01)i, H04R1/28(2006.01)i

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)

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

Published examined utility model applications of Japan 1922-1996
Published unexamined utility model applications of Japan 1971-2007
Registered utility model specifications of Japan 1996-2007
Published registered utility model applications of Japan 1994-2007

Electronic data base consulted during the international search (name of data base and, where practicable, 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	JP 63-135100 A (BIBA KABUSHIKI KAISHA) 1988.06.07, Full Text; Fig.1 (No Family)	1-19
A	JP 2003-143677 A (MITSUBISHI ELECTRIC CORPORATION) 2003.05.16, Full Text; Fig.2 (No Family)	1-19
A	JP 01-253352 A (MATSUSHITA ELECTRIC WORKS,LTD.) 1989.10.09, Full Text; Fig.2 (No Family)	1-19

☐ 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

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"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

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"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
25.12.2007	08.01.2008

Name and mailing address of the ISA/JP	Authorized officer
Japan Patent Office 3-4-3, Kasumigaseki, Chiyoda-ku, Tokyo 100-8915, Japan	Hiroshi Ono 5Z 3654 Telephone No. +81-3-3581-1101 Ext. 3541

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(75) Inventors/Applicants *(for US only)*: **USHER, John** [GB/CA]; 55 de Bresoles, Apt. 400, Montreal, Québec H2Y 1V7 (CA). **BOILLOT, Marc** [US/US]; 9110 NW 11th Court, Plantation, FL 33322 (US).

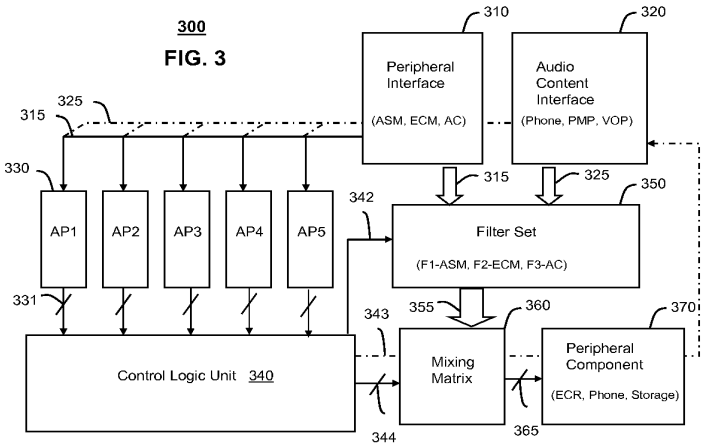
(74) Agents: **KURTZ, Richard** et al.; Greenberg Traurig LLC, Suite 1000, 2101 L Street NW, Washington, DC 20037 (US).

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Published:
— *with international search report*

(54) **Title:** METHOD AND DEVICE FOR LINKING MATRIX CONTROL OF AN EARPIECE II



(57) **Abstract:** A device (100) and method (500) for configuring audio delivery on an earpiece is provided. The earpiece can include a peripheral interface (310) configured to receive a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules (330) that produce audio control information (331) responsive to an analysis of the sound signals, a logic control unit (340) to receive the audio control information from the plurality of audio processing modules and generate configuration data (342), a filter set (350) to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals (355), and a mixing matrix (360) to mix the plurality of filtered sound signals in accordance with the configuration data to produce output sound signals (365) and route the output sound signals to at least one peripheral component (370). Other embodiments are disclosed.

WO 2009/023784 A1

METHOD AND DEVICE FOR LINKING MATRIX CONTROL OF AN EARPIECE II

FIELD

[0001] The present invention relates to a method of controlling an earpiece device, and more particularly, though not exclusively, a method of configuring and managing audio input and output on an earpiece.

BACKGROUND

[0002] Headphones or earpieces can be used for music enjoyment or voice communication. Use of these devices has steadily been increasing, and more products are expanding functionality to support audio delivery to headphones and earpieces.

[0003] The earpieces and associated products are becoming more intelligent as more communication features become available. However, the earpieces themselves have limited resources and must make efficient use of the many audio input and output configurations required to manage and support audio delivery.

[0004] A need therefore exists for improving audio configuration paths of headphones or earpieces.

SUMMARY

[0005] In a first embodiment, an earpiece, can include a peripheral interface configured to receive a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules that produce audio control information responsive to an analysis of the sound signals, a logic control unit operatively coupled to the plurality of audio processing modules to receive the audio control information from the plurality of audio processing modules and generate configuration data, a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals, and a mixing matrix operatively coupled to the control logic unit and filter set to mix the plurality of filtered sound signals in accordance with the audio control information to produce output sound signals and route the output sound signals to at least one peripheral component.

[0006] The peripheral component can be an Ear Canal Receiver (ECR), a phone, a portable communication device, or a data storage. The peripheral interface can include at

least one Ambient Sound Microphone (ASM) configured to convert an ambient sound to an ambient sound signal, and at least one Ear Canal Microphone (ECM) configured to convert an internal sound from an ear canal of a user to an internal sound signal. The earpiece can further include an audio content interface configured to receive a plurality of audio streams and direct the plurality of audio streams to the plurality of audio processing modules. The audio content interface can receive an audio stream from a phone, a media player, or a portable communication device. The audio content interface can mix the plurality of audio streams based on a user context that is one among an incoming call, a music session, or a voice mail.

[0007] The audio control information can include filter data for processing the plurality of sound signals, audio control data for assigning a priority to the plurality of sound signals, and router data for mixing the plurality of filtered signals according to the priority. The priority can be event driven responsive to detecting a sound signature, a background noise condition, a battery life indication, a manual interaction, or a voice recognition command.

[0008] In a second embodiment an earpiece can include a peripheral interface configured to receive a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules, an audio content interface configured to receive a plurality of audio streams and also direct the plurality of audio streams to the plurality of audio processing modules, at least one signal analysis module operatively coupled to the peripheral interface and audio content interface to provide a shared analysis of the sound signals and the audio streams for the plurality of audio processing modules, a logic control unit operatively coupled to the plurality of audio processing modules and the at least one signal analysis module to receive the shared analysis and audio control information to generate configuration data, a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals, and a mixing matrix operatively coupled to the control logic unit and filter set to mix the plurality of filtered sound signals in accordance with the audio control information to produce output sound signals and route the output sound signals to at least one peripheral component.

[0009] The shared analysis can include spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis. A separate signal analysis module can be provided for components of the peripheral interface and the audio content interface that is shared among the plurality of

audio processing modules. The peripheral interface can include at least one Ambient Sound Microphone (ASM) coupled to an ASM signal analysis module configured to analyze an ambient sound signal, and at least one Ear Canal Microphone (ECM) coupled to an ECM signal analysis module configured to analyze an internal sound from an ear canal of a user. The audio content interface can be coupled to an audio content (AC) signal analysis module and configured to analyze an audio stream from a phone, a media player, or a portable communication device. The logic control unit can produce a linkage matrix of mixing gains that are applied to the plurality of filtered sound signals to produce the plurality of output sound signals to each of a respective peripheral output device.

[0010] In a third embodiment, a method for configuring audio delivery on an earpiece can include the steps of receiving at least one sound signal and at least one audio stream, performing an analysis of the at least one sound signal and the at least one audio stream, presenting the analysis, the at least one sound signal, and the at least one audio stream to a plurality of audio processing modules that generate configuration data responsive to the receiving. The method can include filtering the at least one sound signal and at least one audio stream according to the configuration data to produce filtered sound signals, mixing the filtered signals according to the configuration data to produce output sound signals, and routing the output sound signals to at least one peripheral component. The at least one sound signal can be an ambient sound signal, an ear canal sound signal. The at least one audio stream can be received from a phone, a media player, or a portable communication device. A linkage matrix of mixing gains can be generated based on the configuration data that are applied to the plurality of filtered sound signals for producing the plurality of output sound signals.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] FIG. 1 is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

[0012] FIG. 2 is a block diagram of the earpiece in accordance with an exemplary embodiment;

[0013] FIG. 3 is an exemplary schematic of a software system for the earpiece in accordance with an exemplary embodiment;

[0014] FIG. 4 is a more detailed exemplary schematic of the software system of FIG. 3 for the earpiece in accordance with an exemplary embodiment t;

[0015] FIG. 5 is a flowchart of a method for generating filter configuration data from a plurality of processing modules in accordance with an exemplary embodiment in accordance with an exemplary embodiment;

[0016] FIG. 6 is a flowchart of a method for generating and applying filter configuration data from a plurality of processing modules in accordance with an exemplary embodiment;

[0017] FIG. 7 is an exemplary schematic for configuring audio input and output via a mixing matrix in accordance with an exemplary embodiment;

[0018] FIG. 8 is another exemplary schematic of a software system for the earpiece providing separate analysis and re-synthesis modules for peripheral inputs in accordance with an exemplary embodiment; and

[0019] FIG. 9 is a more detailed schematic for analysis and re-synthesis for a particular peripheral input in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

[0020] The following description of at least one exemplary embodiment is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0021] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the relevant art may not be discussed in detail but are intended to be part of the enabling description where appropriate, for example the fabrication and use of transducers. Additionally in at least one exemplary embodiment the sampling rate of the transducers can be varied to pick up pulses of sound, for example less than 50 milliseconds.

[0022] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0023] Note that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed for following figures.

[0024] Note that herein when referring to correcting or preventing an error or damage (*e.g.,* hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0025] At least one exemplary embodiment of the invention is directed to an earpiece for background noise mitigation. Reference is made to **FIG. 1** in which an earpiece device,

generally indicated as earpiece 100, is constructed in accordance with at least one exemplary embodiment. As illustrated, earpiece 100 depicts an electro-acoustical assembly 113 for an in-the-ear acoustic assembly, as it would typically be placed in the ear canal 131 of ear 117 of user 135 The earpiece 100 can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, open-fit device, or any other suitable earpiece type. The earpiece 100 can be partially or fully occluded in the ear canal, and is suitable for use with users having healthy or abnormal auditory functioning.

[0026] Earpiece 100 includes an Ambient Sound Microphone (ASM) 111 to capture ambient sound, an Ear Canal Receiver (ECR) 125 to deliver audio to an ear canal 131, and an Ear Canal Microphone (ECM) 123 to capture internal sounds within the ear canal and also assess a sound exposure level within the ear canal. The earpiece 100 can partially or fully occlude the ear canal 131 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the users ear canal 131, and to form an acoustic seal with the ear canal walls 129 at a location 127 between the entrance to the ear canal and the tympanic membrane (or ear drum) 133 Such a seal is typically achieved by means of a soft and compliant housing of assembly 113. Such a seal is pertinent to the performance of the system in that it creates a closed cavity 131 of approximately 5cc between the in–ear assembly 113 and the tympanic membrane 133. As a result of this seal, the ECR (speaker) 125 is able to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum resulting from the sound field at the entrance to the ear canal. This seal is also the basis for the sound isolating performance of the electro-acoustic assembly.

[0027] Located adjacent to the ECR 125, is the ECM 123, which is acoustically coupled to the (closed) ear canal cavity 131. One of its functions is that of measuring the sound pressure level in the ear canal cavity 131 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of itself and the ECR. The ECM 123 can also be used for capturing voice that is resonant within the ear canal when the user is speaking to permit voice communication.

[0028] The ASM 111 is housed in an ear seal 113 and monitors sound pressure at the entrance to the occluded or partially occluded ear canal. The ASM 111 can also be used to capture the user's voice externally for permitting voice communication. All transducers shown can receive or transmit audio signals to a processor 121 that undertakes audio

signal processing and provides a transceiver for audio or voice via the wired or wireless communication path 119.

[0029] The earpiece 100 can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safes sound reproduction levels. The earpiece 100 in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

[0030] Referring to **FIG. 2**, a block diagram of the earpiece 100 in accordance with an exemplary embodiment is shown. As illustrated, the earpiece 100 can include a processor 206 operatively coupled to the ASM 111, ECR 125, and ECM 123 via one or more Analog to Digital Converters (ADC) 202 and Digital to Analog Converters (DAC) 203. The processor 206 can measure ambient sounds in the environment received at the ASM 111 and internal sounds captured at the ECM 130. Ambient sounds correspond to sounds within the environment such as the sound of traffic noise, street noise, conversation babble, or any other acoustic sound.

[0031] Ambient sounds measured by the ASM 111 can also correspond to industrial sounds present in an industrial setting, such as, factory noise, lifting vehicles, automobiles, and robots. The processor 206 can monitor the ambient sound captured by the ASM 110 for sounds in the environment, such as an abrupt high energy sound corresponding to an on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.) , voice (e.g., “help”, “stop”, ”police”, etc.), or specific noise type (e.g., breaking glass, gunshot, etc.).

[0032] Internal sounds measured by the ECM 123 can correspond to sounds contained within the ear canal 131 such as spoken voice or audio content delivered by way of the ECR 125. The internal sounds can include residual background noise related to ambient sounds in the environment; for example, high level sounds that leak around the ear seal 127 and enter the ear canal 131. The processor 206 can monitor internal sounds captured by the ECM 123 and analyze the internal sounds. The processor 206 can also adjust a mixing between the ambient sound signals measured at the ASM 111 and the internal sound signals measured at the ECM 123, for example, responsive to assessing ambient background noise conditions.

[0033] The processor 206 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated storage memory 208 such a Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device 100. The memory 208 can store program instructions for execution on the processor 206 as well as captured audio processing data.

[0034] The memory 208 can also store program instructions for execution on the processor 206. The memory 208 can be off-chip and external to the processor 208, and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0035] The memory 208 can be a machine-readable medium. The term "machine-readable medium" should be taken to include a single medium or multiple media (e.g., a centralized or distributed, and/or associated caches and external memory) that store the one or more sets of instructions. The term "machine-readable medium" shall also be taken to include any medium that is capable of storing, encoding or carrying a set of instructions for execution by the machine and that cause the machine to perform any one or more of the methodologies of the present disclosure.

[0036] The term "machine-readable medium" shall accordingly be taken to include, but not be limited to: solid-state memories such as a memory card or other package that houses one or more read-only (non-volatile) memories, random access memories, or other re-writable (volatile) memories; and/or magneto-optical or optical medium; and carrier wave signals such as a signal embodying computer instructions in a transmission medium. Accordingly, the disclosure is considered to include any one or more of a machine-readable medium or a distribution medium, as listed herein and including art-recognized equivalents and successor media, in which the software implementations herein are stored.

[0037] The earpiece 100 can include an audio interface 212 operatively coupled to the processor 206 to receive audio content, for example from a media player or cell phone, and deliver the audio content to the processor 206. The processor 206 responsive to detecting

ambient sounds can adjust the audio content and pass the ambient sounds directly to the ear canal. For instance, the processor 206 can lower a volume of the audio content played out the ECR 125 responsive to detecting an acute sound for transmitting the ambient sound to the ear canal. The processor 206 can also actively monitor the sound exposure level inside the ear canal via the ECM 123 and adjust the audio content to within a safe and subjectively optimized listening level range.

[0038] The earpiece 100 can further include a transceiver 204 that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver 204 can also provide support for dynamic downloading over-the-air to the earpiece 100. It should be noted also that next generation access technologies can also be applied to the present disclosure.

[0039] The power supply 210 can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 100 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 210 to improve sensory input via haptic vibration. As an example, the processor 206 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0040] The earpiece 100 can further represent a single operational device or a family of devices configured in a master-slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece 100 can be reused in different form factors for the master and slave devices.

[0041] **FIG. 3** is an exemplary schematic of a software system 300 for operating the earpiece 100. The software system 300 can reside at least in part or whole on the processor 206, the memory 208, and/or any associated machine readable storage medium operated on by the processor 206 (see FIG. 2). Generally stated, the software system 300 by way of the processor 206 can manage a configuration of audio input and output (paths) to the earpiece 100 to support audio delivery.

[0042] As illustrated, the software system 300 includes a peripheral interface 310 configured to manage a plurality of sound signals 315 and direct the plurality of sound signals 315 to a plurality of audio processing modules 330. The sound signals can be an

ambient sound signal measured by the ASM 111 or an internal sound signal from measured by the ECM 123. The software system 300 can include an audio content interface 320 configured to manage a plurality of audio streams 325 and direct the plurality of audio streams 325 to the plurality of audio processing modules 330. The audio stream can be a voice signal from a Phone, a music signal from a personal media player (PMP), or an audio signal provided by the earpiece (e.g., loopback signal). The plurality of sound signals 315 and the plurality of audio streams 325 are also passed to a filter set 350 as shown by the wide arrows.

[0043] The plurality of audio processing modules 330 can produce audio control information 331 responsive to an analysis of the sound signals. The audio control information 331 is provided to the logic control unit 340 for configuring filtering and mixing operations on the plurality of sound signals 315 and the plurality of audio streams 325 at a filter set 350. The logic control 340 unit is operatively coupled to the plurality of audio processing modules 330 to receive the audio control information 331 from the plurality of audio processing modules 330 and generate configuration data. The configuration data can include filter data 342 for processing the plurality of sound signals, audio control data 343 for assigning a priority to the plurality of sound signals, and router data 344 for mixing a plurality of filtered signals 355 according to the priority

[0044] The software system 300 includes the filter set 350 operatively coupled to the peripheral interface to process the plurality of sound signals 315 in accordance with the configuration data 342 for generating the filtered sound signals 355. The filter set 350 also processes the plurality of audio streams 325 in accordance with the configuration data 344 to produce filtered sound signals 355. The filtered sound signals 355 are passed to a mixing matrix 360.

[0045] The mixing matrix 360 is operatively coupled to the control logic unit 340 and filter set 350 to mix the plurality of filtered sound signals 355 in accordance with the configuration data 344 and generate output sound signals 365 that are routed to at least one output peripheral component 370. The peripheral component 370 can be the ECR 125, a phone, or a data storage. The mixing matrix 360 can include individual gains that are applied to the plurality of filtered sound signals 355 for producing the output sound signals.

[0046] **FIG. 4** is a more detailed schematic of the software system 300 of FIG. 3. System components of FIG. 3 will be referred to when describing components of the detailed schematic.

[0047] The peripheral interface 310 (see FIG. 3) can include the Ambient Sound Microphone (ASM) 111 configured to convert an ambient sound to an ambient sound signal, and the Ear Canal Microphone (ECM) 123 configured to convert an internal sound from an ear canal of a user to an internal sound signal. The peripheral interface 310 receives the ASM sound signals from the external environment including background noise, warning sounds, and the ECM sound signal including the voice of a wearer of the earpiece 100 and any audio content playing out the ECR 125. The components (e.g., ASM-R, ECM-R) of the peripheral interface 310 can contain more than the components shown, for instance, a secondary ASM for sound localization or noise suppression. The peripheral interface 310 can also include components (ASM-L, ECM-L) from a second earpiece such as a L-left or R-right earpiece. In this case, the software system 300 can reside on one of the earpieces, although resources can be shared if the software system 300 is enabled on both earpieces.

[0048] The audio content interface 320 (see FIG. 3) can receive audio streams, for example, from a phone 421, a personal media player (PMP) 422, a portable communication device (e.g., VOP) 423, or a component 411 of the earpiece. The local component 411 (e.g., ECR 125) can permit audio feedback for allowing the user to hear audio, for example, loop back when the user is speaking on the phone, or playing comfort noise during non-speech intervals. The audio content interface (AC) 420 can mix the plurality of audio streams to produce a single audio stream delivered to the processing modules (AP1-AP5). In one arrangement, the AC 420 can perform audio content mixing based on a user context.

[0049] The user context identifies an operation or mode of the earpiece, such as, an incoming call, a music session, or a voice mail. For example, if the user is listening to music and an incoming call is detected, the earpiece 100 by way of the software system 300 (see FIG. 3) further discussed herein can lower the volume of the music relative to the incoming call ring tone to permit a hearing of the incoming call (e.g., ring tone). As another example, if the user is using the earpiece in a transparent “safe” mode while listening to music, then any harmful ambient sounds (e.g., loud jackhammer, bus noises) can be attenuated with the music, for instance, by reducing the ASM to ECM pass through levels. Similarly, important warning sounds captured at the ASM 111 can be elevated in the mix with respect to other sounds by increasing the ratio of ASM 111 to ECM 123 sound levels. The AC 420 can also mix the sounds in accordance with manual intervention, for example, if the user

adjusts a volume level manually by way of a user interface or directly mixes the audio streams.

[0050] Referring to both FIGS. 3 and 4, the sound signals from peripheral interface 310 and the mixed sound signals from the audio content interface 320 can be passed through a signal analysis module 410 operatively coupled to the peripheral interface 310 and the audio content interface 320 to provide a shared analysis of the sound signals and the audio streams. The sound signals and the shared analysis can then be sent to the plurality of audio processing modules (AP1 – AP5).The shared analysis can include spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis.

[0051] The shared analysis module 410 permits a sharing of processing resources among the processing modules (AP1 – AP5) to minimize individual resource use of the processing modules from performing a common analysis. For instance, instead of each processing module performing an FFT analysis, the shared module 410 can perform an FFT and share output results with the modules. In another arrangement, output results from the shared analysis can be conveyed to the processing modules based on frequency band requirements or trigger events and thresholds. For instance, AP1 may register as an event listener for ASM signals that exceed a certain level in a frequency band; AP2 may register as an event listener to receive ECM signals that match a particular masking profile, and so on.

[0052] Alternatively, each processing module can interpret the output results individually and make their own respective determination as how to process the sound signals. For instance, each processing module (AP1 – AP5) can selectively process the plurality of sound signals in accordance with its own requirements. For example, AP1 can be a sound detection module to identify one or more sounds in the user's environment, for example, in accordance with the teachings presented in U.S. Patent Application No11/966,457 filed on December 28, 2007 entitled "Method and Device for Sound Signature Detection" herein incorporated by reference in its entirety. AP 2 can be a sound exposure monitoring module to assess safe listening levels, for instance, in accordance with the teachings presented in U.S. provisional patent application No. 60/887,165 filed on January 30, 2007 entitled "Sound Pressure Level Monitoring and Notification System" herein incorporated by reference in its entirety. AP 3 can be a sound enhancement module or voice control module, for instance, in accordance with the teachings presented in U.S. Provisional

Application No. 60/911,691 filed on April 13, 2007 entitled "Method and Device for Voice Operated Control", and U.S. Provisional Application No. 60/885,917 filed on January 22, 2007 entitled "Method and Device for Acute Sound Detection and Reproduction" herein incorporated by reference in its entirety. AP 4 can be a sound correction module to modify a sound signal based on a sound level exposure within safe listening levels, for instance, in accordance with the teachings presented in U.S. Provisional Application No. 60/866,420 filed on November 18, 2006, entitled "Method and Device for Personalized Hearing" the entire disclosure of which is incorporated herein by reference.

[0053] Each of the processing modules (AP1 – AP5) generates separate audio control information 331 in view of the audio signals that can be used by the logic control unit 340 to make an informed decision related to filtering and mixing the sound signals and audio streams. The audio control information 331 includes filter control data 342, audio control data 343, and router control data 344. The filter control data 342 can include filter coefficients that each processing module (AP) identifies as being significant to their intended function. The audio control data 343 can control a mixing of the plurality of audio streams at the AC 420 based on an established priority in view of the processing module decisions (e.g., warning signal detected, sound exposure level exceeded, echo feedback condition). The router control data 344 can include mixing gains (g1-g2) to amplify or attenuate the filtered sound streams.

[0054] The logic control unit 340 responsive to receiving the audio control information 331 generates a priority that is used for generating the configuration data (341,342 and 343) used to filter the sound signals and mix the filtered sound signals. The priority controls how the filtered signals are filtered at the filter set 350 (e.g. F1, F2, F3, see also FIG. 3). As illustrated, F1 is a first filter for the ambient sound signal, and F2 is a second filter for the internal sound signal in the ear canal. The logic control unit 340 varies the level of filtering at both F1 and F2 in accordance with the filter configuration data 342 as established by the determined priority. The filter coefficients for the F1 filter and F2 filter are provided on a frame-by-frame basis in real-time with the filter configuration data 342 from the logic control unit 340. The priority also controls how the audio steams are mixed in the audio control unit 320 (e.g., see VOP, Phone, PMP, see also FIG. 3) based on the audio control data, and how the filtered signals are mixed in the mixing matrix 360 (e.g., g1-g9, see also FIG. 3).

[0055] For example, an AP1 module configured for warning sound detection upon identifying a warning sound in an ASM sound signal, can generate filter coefficients to

amplify one or a group of frequency bands of the warning sound (e.g., horn or siren). The AP1 module can consider the other frequency bands as don't cares since they are not part of the warning sound. AP1 can also raise or flag a priority level that a warning sound has been detected. Similarly, AP3 module for speech enhancement may determine from a simultaneous ECM audio stream that the wearer is speaking and generate filter coefficients that accentuate certain portions of the sound spectrum. AP3 can raise a flag or priority level based on an energy level of the spoken voice. AP4 may determine an echo condition feedback that could potentially damage the users hearing and generate filter coefficients to null out the feedback. AP4 can raise a flag or priority level indicating an immediate danger condition. AP2 may determine that a sound level exposure is being exceeded and generate audio control data to turn volume down on a media player. The logic control unit 340 upon receiving the audio control information and corresponding priorities from the respective processing modules (AP1-AP5) can then determine the appropriate filter configuration given the filter coefficients and the priority.

[0056] Notably, the logic control unit 340 evaluates the audio control information from all the processing modules (AP1-APN) individually, and then collectively as a whole, for prioritizing the configuration data. For instance, warning sounds that are given a higher priority over voice from the ECM or music from the AC, will be mixed according to the priority. Thus, the output sound signal generated by the mixing matrix 360 emphasizes the warning sound, followed by the voice, followed by the audio content. If voice commands are given a higher priority over music, than the mixing matrix 360 can reduce the music levels from the AC module when an AP module detects voice. Again, the priority can be established manually (e.g., via user interface) or automatically (e.g., user context, or presence). As another example, the mixing matrix 360 can increase ASM to ECM pass through upon detection of a warning sound, while simultaneously lowering a phone volume if the user is checking voice mail messages. As yet another example, the mixing matrix 360 can reduce ASM to ECM pass and simultaneously lower a music level if an incoming phone call is detected. The priority can also be event driven responsive to detecting a sound signature, a background noise condition, a battery life indication, a manual interaction, or a voice recognition command.

[0057] In the foregoing, a detailed description for one exemplary operation of audio input/output configuration of the software system 400 shown in FIG. 4 is provided. It should

be noted that other configurations apply, for instance, using more or less than the number of mixing coefficients (g1-g9) or filters (F1-F9) shown.

[0058] As illustrated, The AC control unit 420 mixes audio content (e.g., phone, VOP, PMP) and outputs an AC signal (or pair of stereo channels if two earphone devices are used). The mixing is controlled accordance with configuration data 343 received from the control logic unit 340 and also automatically by the AC control unit 420. The AC signal generated from the AC 420 is then filtered by audio content filter F3. The internal ear-canal audio signal from the ECM 123 is filtered by ECM filter F1 205, and the ambient audio signal from the ASM is filtered by ASM filter F2. All three filters (F1, F2 and F3) are updated and controlled by the logic control unit 340 via filter configuration data 342.

[0059] The filtered audio signals 355 are then passed to the mixing matrix 360. The 9 mixing coefficients (g1-g9) of the mixing matrix 360 are controlled and updated by the control logic unit 340 via configuration data 344. Each mixing coefficient (e.g., g1-g9) can be a time-varying positive number, negative number, or zero. As part of the mixing, the output sound signals are routed to the three peripheral output components (Phone 421, ECR 125, and Storage 430). The mixing gains can be normalized so as to permit balanced audio delivery. The peripheral output components comprise the ear canal receiver ECR 125 for delivering audio to the ear canal, the output signal to the phone 421 (i.e. the signal that is transmitted to another remote individual), and the audio storage device 430 (e.g. hard drive on a PMP).

[0060] The filter configuration data 342 for the 2 filters, the AC configuration data 343 for the AC control unit, and the routing configuration data 344 for the mixing matrix 360 is generated responsive to receiving the audio control information from the plurality of processing modules (AP1-AP5). Recall, each module (e.g., modules 1-5, or more) receives as its inputs at least one of the unfiltered audio input signals (e.g. the ECM and ASM signals) or audio sound streams (e.g., music signal from PMP). As illustrated, each module analyzes the input audio signal and generates 3 output types: coefficients for the router (e.g., g1-g9), filter coefficients for the 3 signal filters (e.g., F1,F2 and F3), and audio control signals for the AC control unit. Each of these output types can have “don’t care” states-which can be represented by a reserved number value (e.g., -1).

[0061] The control logic unit 340 prioritizes the sound signals and audio streams for each signal type, for instance, it can prioritize the router coefficient values (e.g., mixing gains g1-g9) higher for one AP module than the other. In some cases, the control logic can

maximally prioritize the signal type from one module so that the output router coefficients, for example, from one module is copied directly to the output of the control logic unit. In other cases, the output signals of the same type from different modules can be combined by a weighted addition. Similarly, the filter coefficients for a particular filter (F1 – F3) can be generated by a weighted addition (frequency domain) or convolution (time domain).

[0062] **FIG. 5** is a flowchart of a method 500 for generating filter configuration data in accordance with an exemplary embodiment. The method 500 be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method 500, reference will be made to components of FIG 2, although it is understood that the method 500 can be implemented in any other manner using other suitable components. The method 500 can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery device.

[0063] In particular, method 500 describes one exemplary arrangement in which the logic circuit 340 generates filter configuration data from the audio content received by individual AP modules to produce a combination filter output. In the exemplary embodiment, only one filter of the three filters shown in FIG. 4 is explained (e.g. generation of F1 coefficients for the ECM audio path). In other exemplary embodiments, the filter output for method 300 includes all 3 filters (F1, F2 and F3) , for instance, the ECM sound signal, the ASM sound signal and the AC sound signal.

[0064] The method 500 can start at step **502**. At step **504**, the logic control unit 340 can receive N filter vectors from the N processing modules **506** (AP1-APN). Each processing module generates a filter vector that is a set of filter coefficients (time or frequency domain). Each of the N processing modules (AP1-APN) also outputs an associated module priority.

[0065] At step **508**, the logic control unit 340 assesses the priorities raised by each processing module and evaluates the priorities against a corresponding reference module priority at step **510**. If a processing module does not assign a particular priority value, then a default priority value is retrieved from the reference module priority list. At step **512**, the logic control unit 340 identifies the processing module with the highest priority. If a first processing module returns the same module priority value as a second processing module, then the logic control unit 340 assigns a unique module priority value to both the second and first modules, for example, based on a user context.

[0066] Upon selecting the processing module with the highest priority, the logic control unit 340 proceeds to determine if any configuration data is a “don’t care” as shown in step

514. A "don't care" value is a value that does not affect the filter state of the respective processing module. The filter configuration data 342 generated by each processing module can include filter values of a time domain filter vector (e.g. for an FIR type filter) of frequency domain filter (e.g., bi-quads) with "don't care" value; the logic control unit 340 can assign a reserved numerical value of symbol for "don't care" filter states. A "dont care" flag can be associated with each filter output from each module.

[0067] At step **516**, the logic control unit 340 can copy the configuration data (e.g., filter data, router data, audio content data) for "care" states to internal memory for output to the filter set 350, mixing matrix 360, and audio control 420 (see FIG. 4). The logic control unit 340 can continue to update the configuration data for "care" states as it is retrieved from each of the respective processing modules according to the priorities. Starting with the next highest priority module, as shown in step **520**, the filter values for this high priority module are then checked at step **514**to see if they correspond to a "don't care" state in the method loop. The method loop continues until the end of the module priority list as shown in step **518**.

[0068] If the logic control unit 340 identifies a "don't care" state is present, the filter values for that state is not updated; else, the filter values for this module are copied to the output and no further update of the filters is conducted until new filter vectors are received from all modules as shown in step **522**. In such regard, the logic control unit 340 output filter states using filter values directly from the processing modules based strictly on priority. If all module filter values are "don't care" states, then the filter output is the same as a previous filter output. The method 500 continues back at step **504** to evaluate the priorities of each module in real-time as results are received. The logic control unit 340 thus continually updates the filter values in view of priority for the respective filters (F1, F2 and F3) to process audio signals and audio streams in real time.

[0069] **FIG. 6** is a flowchart of a method 600 for generating filter configuration data where filter data is collectively combined based on priority. The method 600 is an extension of method 500 of FIG. 5 and can practiced with more or less than the number of steps shown and is not limited to the order shown. Briefly, FIG. 6 presents an embodiment wherein processing modules can have a same priority value. In such a case when two modules have the same priority value, and neither contain (don't care" filter states, then the filter values from each of these two module with the same priority value are combined. The

combination can be a simple summation of filter weights (e.g. for frequency domain filters) or a convolution of the two sequences (e.g. for time domain weights).

[0070] The method 600 can start at step **602**. At step **604**, the logic control unit 340 can receive N filter vectors from the N processing modules **606** (AP1-APN). Each processing module generates a filter vector that is a set of filter coefficients (time or frequency domain). Each of the N processing modules (AP1-APN) also outputs an associated module priority. At step **608**, the logic control unit 340 assesses the priorities raised by each processing module. If a processing module does not assign a particular priority value, then a default priority value can be retrieved from a reference module priority list.

[0071] At step **610**, the logic control unit 340 identifies the processing module with the highest priority. The logic control unit 340 at step **618** then examines the audio control information from each of the N processing modules to identify "don't care" states. Filter values with "don't care" are not considered in generating output filter configuration data 342 or router configuration data 344 (see FIG. 4). Accordingly, the method 600 continues to get the next most important priority modules at step **620** until the end of module priority list is reached at step **622**.

[0072] Filter values other than "don't care" are however used in generating a combined filter output; that is, a filter output representative of the collective priority of all the processing modules (AP1-APN) based on their processing results. The combination also depends on whether the module is uniquely defined; for instance, it has precedence over combination with other filter vales. Thus at step **614**, the logic control unit 340 determines whether a processing module with a "care" filter value is uniquely defined. If yes, then the filter values for that unique model are copied to the output (e.g., router configuration, filter configuration data, audio configuration data) at step **616**. If however, the module is not unique, then the logic control unit 340 at step **612** combines the filter value with filter data from modules of the same priority. These filter values are then copied to the output at step **616** if no other modules eclipse their collective priority, or until a module with higher priority is identified. Upon evaluating all new filter from all the modules at step **624**, the method 600 can return back to step 604 to continue to retrieve the filter vectors in real-time.

[0073] FIG. 7 is an exemplary schematic 700 for configuring audio input and output via a mixing matrix in accordance with an exemplary embodiment. Reference will be made to FIG. 4 when describing the schematic 700. For instance, the filtered ECM signal 702 is the output of the ECM filter (F1) of FIG. 4. The mixing gains g1 (704), g4 (704) and g7 (707)

correspond to the first column of the mixing matrix 360 as shown in FIG. 4. The phone out 710, ECR 712, and audio storage 714 correspond to the output peripheral components (421, 125, and 430) of the mixing matrix 360.

[0074] The mixing gains (g1-g9) of the mixing matrix 360 can be applied to the filtered audio signals 355 (see FIG. 3) before they are routed to the corresponding peripheral output components (phone out 710, ECR 712, and audio storage 714). FIG. 7 shows a signal path diagram for the filtered ECM input signal 702 to the 3 peripheral output components of the matrix 260. The mixing gains can be normalized coefficients whose total contribution across a column of the mixing matrix is unity. For example, mixing gains of g1=0.2, g4=0.3, and g7=0.5 are applied to the filtered ECM signal 702 to produce phone out 710, ECR out 125, and audio storage out 714. The coefficient values can be time variant for each new input audio sample into the mixing matrix 360, or can remain constant for a block of input samples.

[0075] In one exemplary embodiment, the 9 coefficients (g1-g9) for each of the mixing gains in mixing matrix 360 can be generated using a similar logic process as described in FIG. 4 and FIG. 4, except the inputs to the logic system are not filter vectors but rather 9 coefficient values from each module. The priority value assigned to each module, or the priority value automatically generated by that module, can be a different value for the filter vectors than for the 9 coefficient values. Moreover, 9 coefficients are shown since there are 3 audio paths (ASM, ECM, and AC) and 3 output peripheral components (Phone, ECR, data storage). The number of mixing gains and structure of the mixing matrix 360 can thus be a function of the number of audio input and audio output paths.

[0076] FIG. 8 is another exemplary schematic 800 of the software system 300 providing separate analysis and re-synthesis modules for peripheral inputs in accordance with an exemplary embodiment. As illustrated, the schematic 800 combines the three audio filters F1, F2 and F3 with the shared analysis module 410 of the software system shown in FIG 4. Analysis/Re-synthesis unit 810 performs shared analysis of the ECM sound signal for processing modules (AP1-AP3). Analysis/Re-synthesis unit 820 performs shared analysis of the ASN sound signal for processing modules (AP1-AP3). Analysis/Re-synthesis unit 830 performs shared analysis of the Audio Content (AC) sound signal for processing modules (AP1-AP3).

[0077] The combined analysis/resysnthesis units 810, 820 and 830 filter the 3 input audio signals; the ECM signal; the ASM signal, and Audio Content (AC) signal. The AC

signal comprises at least one of; an earcon signal (e.g. a “low battery” auditory warning cue); a signal from a mobile telephone; and a stereo signal from a PMP (e.g. portable DVD player, music audio player etc). The output signal of the analysis/resynthesis units 810, 820 and 830 are passed to the mixing columns (g1, g4, g7; g2, g5, g8; and g3, g6, g9) of the mixing matrix 360. The mixing gains g1-g9 are controlled via the router configuration data from the logic control unit 340, which in turn were determined from the audio control information generated by the processing modules (AP1-AP3).

[0078] As illustrated the mixing matrix 360 can associate a phone with a first row of a linkage matrix, associates an ECR with a second row of a linkage matrix, associate an audio storage with a third row of a linkage matrix, associate an ECM with a first column of a linkage matrix, associate an ASM with a second column of a linkage matrix, and associate an AC controller with a third column of a linkage matrix. The mixing matrix can modify the output sound signal of the phone using values in the first row of the linkage matrix, modify the output sound signal of the ECR using values in the second row of the linkage matrix, and modify the output sound signal sent to the audio storage using values in the third row of the linkage matrix.

[0079] Recall each processing module has two or more output signals: filter gain coefficients which control the resynthesis of the signals in the analysis/resythesis units; 9 coefficients for the router, and audio configuration data for the audio control. The processing modules (AP1-AP3) create AC Control signals to control the routing of the input AC signals using AC control unit (420, see FIG. 4). The processing modules are configured and their internal state can be queried by the user interface 850 (e.g. a computer connected to the DSP unit via USB).

[0080] FIG. 9 is a more detailed schematic 900 of an analysis and re-synthesis unit for a particular peripheral input in accordance with an exemplary embodiment. For instance, the analysis and re-synthesis unit 910 corresponds to the ASM sound signal path of the software system 800 shown in FIG. 8. That is, the analysis and re-synthesis unit 910 can receive audio input 901 from the ASM 111. Notably, the schematic 900 can be applied for producing filtered output for also the ECM 123 sound signals and AC 420 sound signals (see FIG. 4) or other signals.

[0081] As illustrated in FIG. 9, the analysis and re-synthesis unit 910 incorporates a filter-bank 912 (e.g., cascade of band-pass filters (BPF)) structure for decomposing the ASM sound signal into a plurality of filter-banks. The filter-bank can be based on a linear

frequency scale or a non-linear frequency scale, such as, an octave, 1/3 octave, critical band, equivalent rectangular band (ERB), or melody (mel)-frequency scale. Each output of each filter-bank can then be presented to each of the N processing modules (AP1 920-AP2 930). As previously indicated, the processing modules 920 and 930 generate audio control information 331 (see FIG. 3) that the logic control unit 340 uses in generating filter configuration data 342 (see FIG. 4). In the embodiment shown, the filter configuration data 342 is used by the analysis and re-synthesis unit 910 to scale the filter-bank output signals. For instance, the filter-bank outputs are gain scaled to amplify or attenuate the individual filter-banks with the filter coefficients in the filter configuration data 342. The filter-bank output signals can then be summed to generate a composite filtered sound signal. The filtered sound signal is then presented to the mixing matrix (see FIG. 8) for routing and mixing to the one or more output peripheral components (e.g., Phone, ECR, data storage).

[0082] In the following, a detailed description for operation of the analysis and re-synthesis unit 910 for use with an audio input signal is provided. The input signal 901 is called “audio input 1”, and can be any one of the exemplary three signals described previously (i.e. ECM, ASM or AC signal).The audio input signal is split and processed by the different band-pass-filters 912 (i.e. with different centre frequencies). In one arrangement, the response of each respective band pass filter 912 is such that if the filter outputs are summed, then the power spectral density is the same as the original audio input signal. The band pass filters 912 can be time-domain FIR type filters or IIR type filters, or the analysis could be with a frequency domain FFT. The output signal of each band-pass filter is fed to at least one module (for the sake of clarity, 2 modules are shown in the exemplary embodiment of figure 8). Each module then creates an output signal to control the re-sythesis of the band-pass filtered input signals. This output signal is a vector of weights to control the gains of the filtered signals (i.e. the number of gains is equal to the number of filter used in the analysis). The logic control unit 340 determines a single set of gain coefficients for the gains (i.e. G1, G2, G3 and G4 in the exemplary embodiment of figure 8). After the first set of band pass filtered signals have been multiplied with the gains (G1-G4) to create a second set of amplified (or attenuated) band-pass filtered signal, the second set of signals are summed with the summing unit to create a single filtered ASM output sound signal, which form one of the three exemplary inputs to the previously described mixing matrix 360.

[0083] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

CLAIMS

What is claimed is,

1. An earpiece, comprising:

a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules that produce audio control information responsive to an analysis of the plurality of sound signals;

a logic control unit operatively coupled to the plurality of audio processing modules to receive the audio control information from the plurality of audio processing modules and generate configuration data;

a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals; and

a mixing matrix operatively coupled to the control logic unit and filter set to mix the plurality of filtered sound signals in accordance with the configuration data to produce output sound signals and route the output sound signals to at least one peripheral component.

2. The earpiece of claim 1, wherein the peripheral component is an Ear Canal Receiver (ECR), a phone, a portable communication device, or a storage.

3. The earpiece of claim 1, wherein the peripheral interface comprises:

at least one Ambient Sound Microphone (ASM) configured to convert an ambient sound to an ambient sound signal; and

at least one Ear Canal Microphone (ECM) configured to convert an internal sound from an ear canal of a user to an internal sound signal.

4. The earpiece of claim 1, further comprising:

an audio content interface configured to manage a plurality of audio streams and direct the plurality of audio streams to the plurality of audio processing modules.

5. The earpiece of claim 4, wherein the audio content interface receives an audio stream from a phone, a media player, or a portable communication device.

6. The earpiece of claim 4, wherein the audio content interface mixes the plurality of audio streams based on a user context that is one among an incoming call, a music session, or a voice mail.

7. The earpiece of claim 1, wherein the configuration data comprises filter data for processing the plurality of sound signals, audio control data for assigning a priority to the plurality of sound signals, and router data for mixing the plurality of filtered signals according to the priority

8. The earpiece of claim 4, wherein the priority is event driven responsive to detecting a sound signature, a background noise condition, a battery life indication, a manual interaction, or a voice recognition command.

9. The earpiece of claim 3, wherein the filter set comprises:

a first filter for the ambient sound signal; and

a second filter for the internal sound signal,

where the filter coefficients for the first filter and second filter are provided on a frame-by-frame basis in real-time with the configuration data from the logic control unit.

10. An earpiece, comprising:

a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules;

an audio content interface configured to manage a plurality of audio streams and direct the plurality of audio streams to the plurality of audio processing modules;

at least one signal analysis module operatively coupled to the peripheral interface and audio content interface to provide a shared analysis of the sound signals and the audio streams for the plurality of audio processing modules;

a logic control unit operatively coupled to the plurality of audio processing modules and the at least one signal analysis module to receive the shared analysis and audio control information to generate configuration data;

a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals; and

a mixing matrix operatively coupled to the control logic unit and filter set to mix the plurality of filtered sound signals in accordance with the audio control information to produce output sound signals and route the output sound signals to at least one peripheral component.

11. The earpiece of claim 10, wherein the shared analysis comprises spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis.

12. The earpiece of claim 10, wherein a separate signal analysis module is provided for components of the peripheral interface and the audio content interface and is shared among the plurality of audio processing modules.

13. The earpiece of claim 10, wherein the peripheral interface comprises

at least one Ambient Sound Microphone (ASM) coupled to an ASM signal analysis module configured to analyze an ambient sound signal; and

at least one Ear Canal Microphone (ECM) coupled to an ECM signal analysis module configured to analyze an internal sound from an ear canal of a user.

14. The earpiece of claim 10, wherein the audio content interface is coupled to an audio content (AC) signal analysis module and configured to analyze an audio stream from a phone, a media player, or a portable communication device.

15. The earpiece of claim 10, wherein the logic control unit produces a linkage matrix of mixing gains that the mixing matrix applies to the plurality of filtered sound signals for producing the plurality of output sound signals.

16. The earpiece of claim 10, wherein the mixing matrix:

associates a phone with a first row of a linkage matrix;

associates an ECR with a second row of a linkage matrix;

associates an audio storage with a third row of a linkage matrix;

associates an ECM with a first column of a linkage matrix;

associates an ASM with a second column of a linkage matrix; and

associates an AC controller with a third column of a linkage matrix.

17. The earpiece of claim 16, wherein the mixing matrix:

modifies the output sound signal of the phone using values in the first row of the linkage matrix;

modifies the output sound signal of the ECR using values in the second row of the linkage matrix; and

modifies the output sound signal sent to the audio storage using values in the third row of the linkage matrix.

18. A method for configuring audio delivery on an earpiece, the method comprising the steps of:

receiving at least one sound signal and at least one audio stream;

performing an analysis of at least one sound signal and the at least one audio stream;

presenting the analysis, the at least one sound signal, and the at least one audio stream to a plurality of audio processing modules that generate configuration data responsive to the receiving;

filtering the at least one sound signal and at least one audio stream according to the configuration data to produce filtered sound signals;

mixing the filtered signals according to the configuration data to produce output sound signals; and

routing the output sound signals to at least one peripheral component.

19. The method of claim 18, wherein the at least one sound signal is an ambient sound signal, an ear canal sound signal, and the at least one audio stream is received from a phone, a media player, or a portable communication device.

20. The method of claim 18, comprising generating a linkage matrix of mixing gains based on the configuration data that are applied to the plurality of filtered sound signals for producing the plurality of output sound signals.

FIG. 1

Ambient Sound
111
ASM
ADC 202
Transceiver 204
Processor 206
Memory 208
Power Supply 210
Audio Interface 212
ADC 202
123
ECM
DAC 203
Audio
ECR
125
Audio Content (e.g., music, cell phone, voice mail)
100

FIG. 2

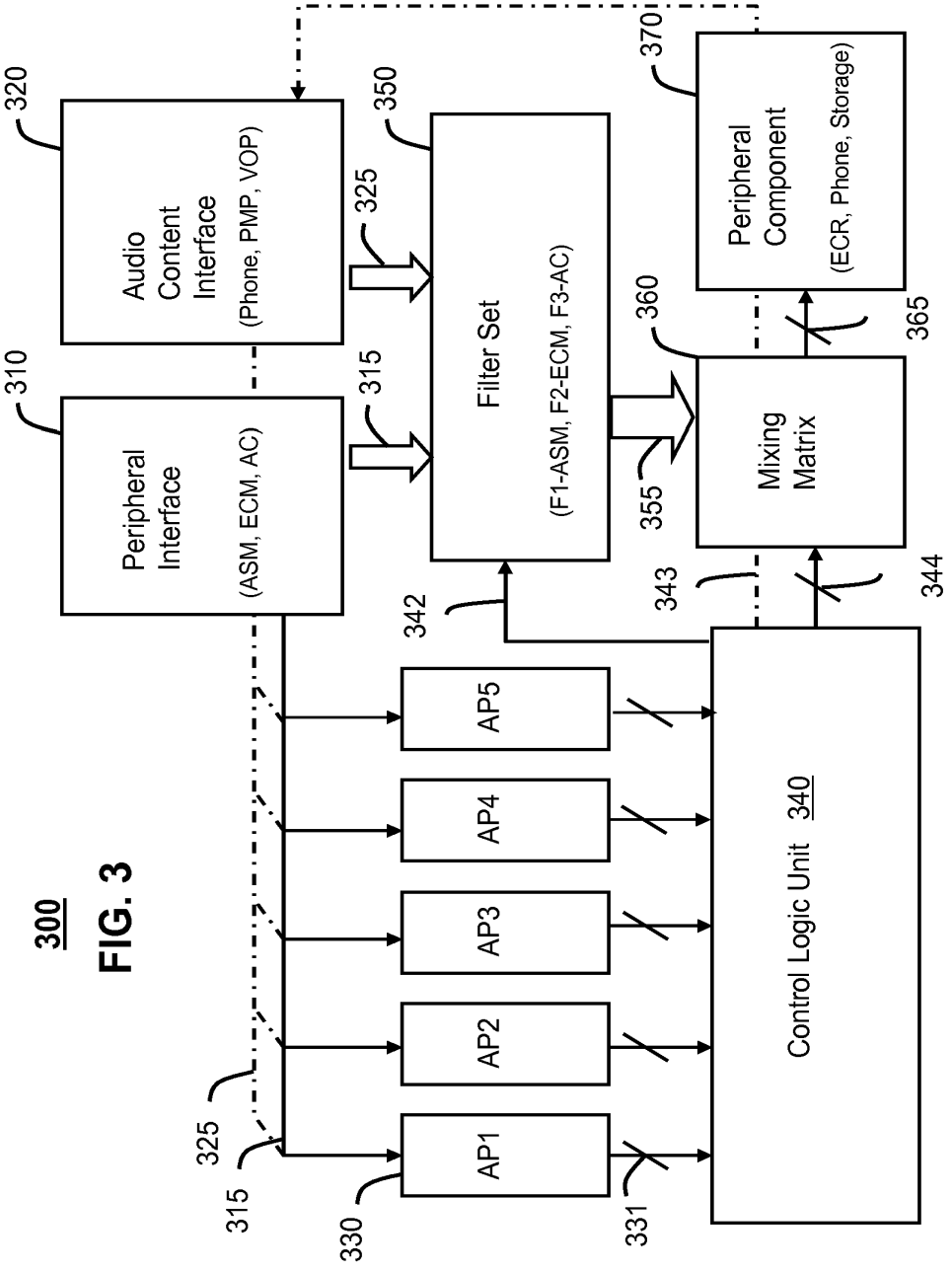
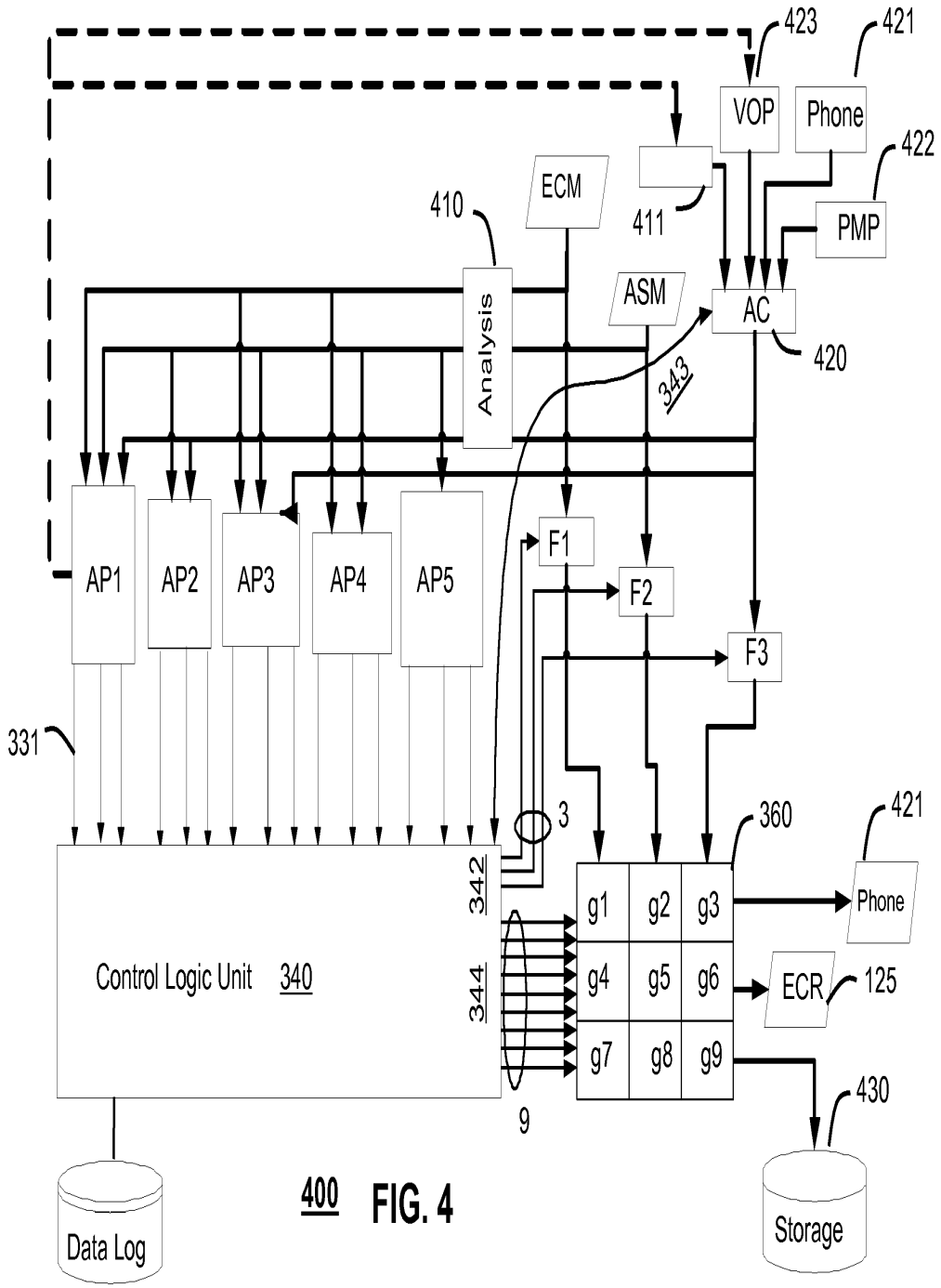

300
320
Audio Content Interface (Phone, PMP, VOP)
310
Peripheral Interface (ASM, ECM, AC)
325
315
350
Filter Set (F1-ASM, F2-ECM, F3-AC)
360
355
Mixing Matrix
370
Peripheral Component (ECR, Phone, Storage)
365
342
343
344
AP5
AP4
AP3
AP2
AP1
Control Logic Unit 340
325
315
330
331

FIG. 3


423
421
VOP
Phone
422
ECM
411
410
PMP
ASM
AC
Analysis
420
343
AP1
AP2
AP3
AP4
AP5
F1
F2
F3
331
3
360
421
342
g1
g2
g3
Phone
Control Logic Unit 340
344
g4
g5
g6
ECR
125
g7
g8
g9
430
9
Data Log
Storage

400 FIG. 4

Start
502
506
Module 1
Module 2
Module N
Get N filter vectors from N modules
504
508
510
Get module priority list
Reference Module priority list
Get highest priority module number (M)
512
500
M
Y
514
516
520
Module M filter "Don't Care"?
N
Copy Module 1 filters to output of Logic Circuit
Get next most important module (module M)
Y
518
End of Module Priority list?
N
522
Y
New filters from all modules?
NO

FIG. 5

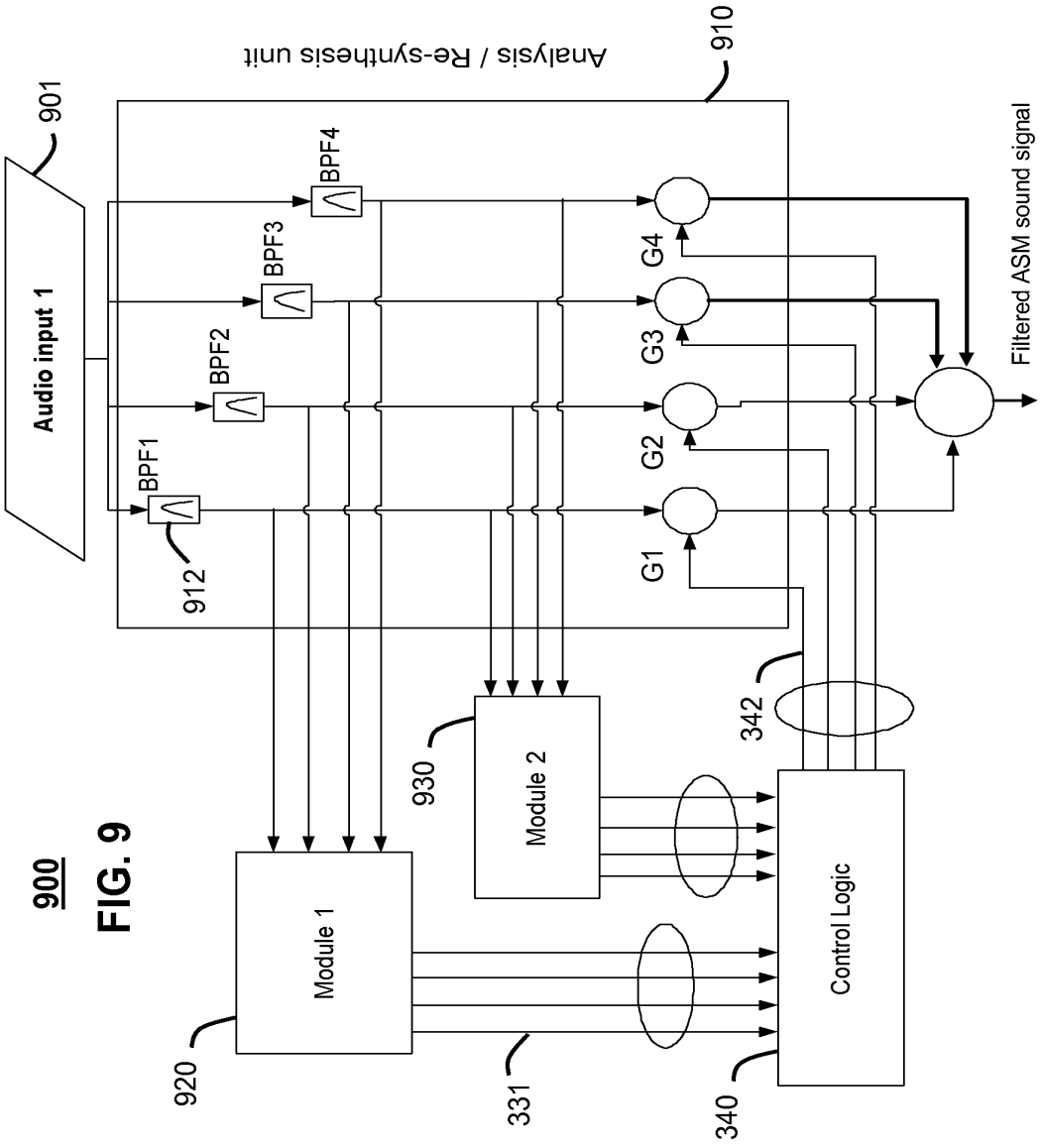
600
FIG. 6
Start
602
606
Module 1
604
Get N filter vectors from N modules
Module N
608
Get module priority list
610
Get highest priority module number(M)
612
Combine filter from the modules with same priorities
N
614
616
620
618
Get next most important priority module(s)
Module M filter "Don't Care?
N
Unique module priority value?
Y
Copy Module1 filters to output of Logic Circuit
Y
622
End of Module Priority list?
N
Y
624
New filters from all modules?
NO

701
g1
704
g4
706
g7
702
Filtered ECM signal
X
710
PhoneOut
712
ECR
714
Audio storeage
700

FIG. 7

VOP
Phone
ECM
AC
PMP
ASM
Analysis/ Resynthesis 810
Analysis/ Resynthesis 820
Analysis/ Resynthesis 830
123
AP1
AP2
AP3
User Interface 850
Control Logic Unit
123
g1
g2
g3
g4
g5
g6
g7
g8
g9
Phone
ECR
Storage
800

FIG. 8


900
FIG. 9
Analysis / Re-synthesis unit
910
901
Audio input 1
BPF1
BPF2
BPF3
BPF4
912
G1
G2
G3
G4
Filtered ASM sound signal
342
930
Module 2
Module 1
920
331
340
Control Logic

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US 08/73189

A. CLASSIFICATION OF SUBJECT MATTER
IPC(8) - H04R 25/00 (2008.04)
USPC - 381/320
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)
USPC: 381/320

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
USPC: 381/312-331 (see terms below)

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
PubWEST(PGPB,USPT,USOC,EPAB,JPAB), GOOGLE SCHOLAR
terms: mix, matrix, linkage, spectral analysis, ambient, environmental, signal, sound, ear canal, meatus, in-the-ear, earpiece, earplug, earbud, priority, cellular phone.

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X ---- Y	US 2005/0058313 A1 (Victorian et al.) 17 March 2005 (17.03.2005), fig. 1, 4a, para [0004]-[0005], [0030]-[0032], [0042]-[0052], [0063].	1-8, 10, 12-20 ---------- 9, 11
Y	US 6,754,359 B1 (Svean et al.) 22 June 2004 (22.06.2004), fig. 1-2, col 4, ln 39 - col 5, ln 3, col 7, ln 26 - col 8, ln 17.	9
Y	US 2006/0262944 A1 (Rasmussen et al.) 23 November 2006 (23.11.2006), fig. 1 and 4, para [0008], [0021]-[0023].	11

☐ Further documents are listed in the continuation of Box C. ☐

* 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
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"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
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Name and mailing address of the ISA/US Mail Stop PCT, Attn: ISA/US, Commissioner for Patents P.O. Box 1450, Alexandria, Virginia 22313-1450 Facsimile No. 571-273-3201	Authorized officer: Lee W. Young PCT Helpdesk: 571-272-4300 PCT OSP: 571-272-7774

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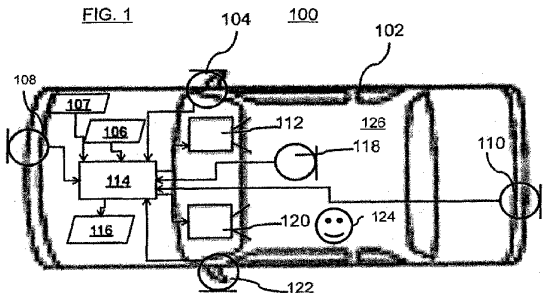
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(54) **Title**: AUTOMOTIVE SOUND RECOGNITION SYSTEM FOR ENHANCED SITUATION AWARENESS

FIG. 1



(57) **Abstract**: Sound recognition systems for a vehicle and methods for increasing auditory situation awareness in a vehicle are provided. A sound recognition system includes at least one ambient sound microphone (ASM), at least one vehicle cabin receiver (VCR) and a processor. The ASM is disposed on the vehicle and configured to capture ambient sound external to the vehicle. The VCR is configured to deliver audio content to a vehicle cabin of the vehicle. The processor coupled to the at least one ASM and the at least one VCR. The processor is configured to detect at least one sound signature in the ambient sound and to adjust the audio content delivered to the vehicle cabin based on the detected at least one sound signature.

WO 2012/097150 A1

AUTOMOTIVE SOUND RECOGNITION SYSTEM FOR ENHANCED SITUATION AWARENESS

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is related to and claims the benefit of U.S. Provisional Application No. 61/432,016 entitled "AUTOMOTIVE SOUND RECOGNITION SYSTEM FOR ENHANCED SITUATION AWARENESS" filed on January 12, 2011, the contents of which are incorporated herein by reference.

FIELD OF THE INVENTION

[0002] The present invention relates to a device that monitors sound directed to a vehicle cabin, and more particularly, though not exclusively, to an audio system and method that detects ambient warning sounds and adjusts audio delivered to a vehicle cabin based on the detected warning sounds to enhance auditory situation awareness.

BACKGROUND OF THE INVENTION

[0003] People that use audio systems in vehicles generally do so for either music enjoyment or voice communication. The user is generally immersed in the audio experience when using such devices and is acoustically isolated within a sealed vehicle cabin. Background noises in the external vehicle environment (e.g., road, engine, wind and traffic noise) can contend with the acoustic sounds produced from these devices. As the background noise levels change, the user may need to adjust the volume to listen to their music over the background noise. Alternatively, the level of reproduced audio may be automatically increased, for example, by audio systems that increase the audio level as the vehicle velocity increases (i.e., to compensate for the rise in noise level from road, engine and aerodynamic noise). One example of such an automatic gain control system is described in US patent No. 5,081,682.

SUMMARY OF THE INVENTION

[0004] Aspects of the present invention relate to a sound recognition system for a vehicle. A sound recognition system includes at least one ambient sound microphone (ASM), at least one vehicle cabin receiver (VCR) and a processor. The ASM is disposed on the vehicle and configured to capture ambient sound external to the vehicle. The VCR is configured to deliver audio content to a vehicle cabin of the vehicle. The processor coupled to the at least one ASM and the at least one VCR. The processor is configured to detect at least one sound signature in the ambient sound and to adjust the audio content delivered to the vehicle cabin based on the detected at least one sound signature.

[0005] Aspects of the present invention also relate to methods for increasing auditory situation awareness in a vehicle. The method includes capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle; monitoring the ambient sound for a target sound by detecting a sound signature

corresponding to the target sound in the ambient sound; and adjusting a delivery of audio content by at least one vehicle cabin receiver (VCR) to a vehicle cabin of the vehicle based on the target sound.

[0006] Aspects of the present invention further relate to methods for sound signature detection for a vehicle. The method includes capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle; and receiving a directive to learn a sound signature within the ambient sound. A voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.

[0007] Aspects of the present invention also relate to methods for personalized listening in a vehicle. The method includes capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle; detecting a sound signature within the ambient sound that is associated with a warning sound; and mixing the warning sound with audio content delivered to the vehicle cabin via at least one vehicle cabin receiver (VCR) in accordance with a priority of the warning sound and a personalized hearing level (PHL).

BRIEF DESCRIPTION OF THE DRAWINGS

[0008] The invention may be understood from the following detailed description when read in connection with the accompanying drawing. It is emphasized, according to common practice, that various features of the drawings may not be drawn to scale. On the contrary, the dimensions of the various features may be arbitrarily expanded or reduced for clarity. Moreover, in the drawing, common numerical references are used to represent like features. Included in the drawing are the following figures:

[0009] FIG. 1 is a pictorial diagram of a vehicle including an exemplary automotive sound recognition system for enhanced situation awareness in accordance with an embodiment of the present invention;

[0010] FIG. 2 is a block diagram of the system shown in FIG. 1 in accordance with an exemplary embodiment of the present invention;

[0011] FIG. 3 is a flowchart of an exemplary method for ambient sound monitoring and warning detection in accordance with an embodiment of the present invention;

[0012] FIG. 4 illustrates various system modes in accordance with an exemplary embodiment of the present invention;

[0013] FIG. 5 is a flowchart of an exemplary method for sound signature detection in accordance with an embodiment of the present invention;

[0014] FIG. 6 is a flowchart of an exemplary method for managing audio delivery based on detected sound signatures in accordance with an embodiment of the present invention;

[0015] FIG. 7 is a flowchart of an exemplary method for sound signature detection in accordance with an embodiment of the present invention;

[0016] FIG. 8 is a pictorial diagram for mixing ambient sounds and warning sounds with audio content in accordance with an exemplary embodiment of the present invention; and

[0017] FIG. 9 is a flowchart of an exemplary method for updating the sound signature detection library dependent on the vehicle location in accordance with an embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

[0018] The following description of at least one exemplary embodiment is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0019] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the relevant art may not be discussed in detail but are intended to be part of the enabling description where appropriate, for example the fabrication and use of transducers. Additionally in at least one exemplary embodiment the sampling rate of the transducers can be varied to pick up pulses of sound, for example less than 50 milliseconds.

[0020] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0021] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0022] Automotive vehicle operators are often auditorially removed from their external ambient environment. Ambient sound cues such as from oncoming emergency (and non-emergency) vehicle sound alerts are often not heard by the vehicle operator due to acoustic isolation of the vehicle cabin and internal cabin noise from engine and road noise and, especially, due to loud music and speech reproduction levels in the vehicle cabin.

[0023] Accordingly, background noises in the external vehicle environment contend with the acoustic sounds produced from the vehicle audio system. The vehicle operator, thus, becomes auditorially disassociated with their ambient environment, thereby increasing the danger of accidents from collisions with oncoming vehicles. A need therefore exists for improving the auditory situation awareness of vehicle operators to automatically alert the operator to ambient warning alerts.

[0024] Music and speech audio reproduction levels in vehicles and ambient sound levels are antagonistic. For example, vehicle operators typically play vehicle audio devices louder to hear over the traffic and general urban noise. The same applies to voice communication.

[0025] Automotive vehicle operators are often auditorially removed from their external ambient environment. For example, high sound isolation from the external environment by be provided by cabin structural insulation, close-fitting window seals and thick or double paned glass. Ambient sound cues (from external acoustic signals), such as oncoming emergency (and non-emergency) vehicle sound alerts; vocal messages from pedestrians; and sounds generated by the operator's own vehicle may often not be heard by the vehicle operator.

[0026] To summarize, the reduced "situation awareness" of the vehicle operator may be a consequence of at least two principle factors. One factor includes acoustic isolation of the vehicle cabin (e.g., from the vehicle windows and structural isolation). A second factor includes sound masking. The sound masking may include masking from internal cabin noise (such as from engine and road noise) and masking from loud music reproduction levels within the vehicle. The masking effect may be further compounded with telephone communications, where the vehicles operator's attention may be further distracted by the conversation. Telephone conversation, thus, may introduce an additional cognitive load that may further reduces the vehicle operator's auditory situation awareness of the vehicle surroundings.

[0027] The reduction of the situation awareness of the vehicle operator may lead to danger. For example, a personal safety of the vehicle operator may be reduced. In addition, personal safety of other vehicle operators and pedestrians in the vicinity of the vehicle may also be threatened.

[0028] One definition of situation awareness includes "the perception of elements in the environment within a volume of time and space, the comprehension of their meaning, and the projection of their status in the near future". While some definitions are specific to the environment from which they were adapted, the above definition may be applicable across multiple task domains from visual to auditory modalities.

[0029] A method and system is herein disclosed to address this problem of reduced auditory situation awareness of vehicle operators. In an exemplary embodiment, ambient warning sounds in the vicinity of a vehicle may be automatically detected and may be actively reproduced in the vehicle cabin, to inform the vehicle operator of detected sounds. According to an exemplary embodiment, a library of known warning sounds may be acquired automatically based on the vehicle's location.

[0030] Personal safety of the vehicle operator and passengers in the vehicle may thereby enhanced by exemplary systems and methods of the present invention, as described herein. Accordingly, the safety of other vehicles (such as oncoming emergency vehicles, other motorists, and pedestrians) may also be increased. The safety benefit comes not only from the enhanced auditory situation awareness, but also via reduced driver workload. For example, the system may reduce the burden on the driver to constantly visually scan the environment for emergency vehicles or other dangers that may also have recognizable acoustical signatures (that may ordinarily be inaudible inside the vehicle cabin).

[0031] One focus of the present invention is to enhance (i.e., increase) the auditory situation awareness of a typical vehicle operator and, thereby, improve the personal safety of the vehicle operator, and other motorists and pedestrians.

[0032] Referring to FIG. 1, a pictorial diagram of vehicle 102 including an exemplary automotive sound recognition system (ASRS) 100 for enhanced auditory situation awareness is shown. ASRS 100 may include user interface 106, central audio processor system 114 (also referred to herein as processor 114), indicator 116 and at least one loudspeaker (for example, right loudspeaker 112 and left loudspeaker 120) (also referred to herein as vehicle cabin receivers (VCRs) 112, 120). ASRS 100 may also include one or more ambient microphones (for example, right microphone 104, front microphone 108, rear microphone 110 and left microphone 122) for capturing ambient sound external to vehicle 102. ASRS 100 may also include at least one vehicle cabin microphone (VCM) 118 for capturing sound within vehicle cabin 126.

[0033] Processor 114 may be coupled to one or more of user interface 106, indicator 116, VCRs 112, 120, VCM 118 and ambient microphones 104, 108, 110, 122. Processor 114 may be configured to control acquisition of ambient sound signals from ambient microphones 104, 108, 110, 122 and (optionally) a cabin sound signal from VCM 118. Processor 114 may be configured analyze ambient and/or cabin sound signals, and to present information by system 100 to vehicle operator 124 (such as via VCRs 112, 120 and/or indicator 116) responsive to the analysis.

[0034] In operation, processor 114 may be configured to receive AC signal 107 and reproduce AC signal 107 through VCRs 112, 120 into vehicle cabin 126. Processor 114 may also be configured to receive ambient sound signals from respective ambient microphones 104, 108, 110, 122. Processor 114 may also be configured to receive a cabin sound signal from VCM 118.

[0035] Based on an analysis of the ambient sound signals (and, optionally, the cabin sound signal), processor 114 may mix the ambient sound signal from at least one of ambient microphones 104, 108, 110, 122 with AC signal 107. The mixed signal may be

output to VCRs 112, 120. Accordingly, acoustic cues in the ambient signal (such as an ambulance siren, a vocal warning from a pedestrian, or a vehicle malfunction sound) may be passed into vehicle cabin 126, thereby providing detectable and spatial localization cues for vehicle operator 124.

[0036] AC signal 107 may include any audio signal provided to (and/or generated by) processor 114 that may be reproduced through VCRs 112, 120. AC signal 107 may correspond to (without being limited to) at least one of the following exemplary signals: a music or voice audio signal from a music audio source (for example, a radio, a portable media player, a computing device); voice audio (for example, from a telephone, a radio device or an occupant of vehicle 102); or an audio warning signal automatically generated by vehicle 102 (for example, in response to a backup proximity sensor, an unbelted passenger restraint, an engine malfunction condition, or other audio alert signals). AC signal 107 may be manually selected by vehicle operator 124 (for example, with user interface 106), or may be automatically generated by vehicle 102 (for example, by processor 114).

[0037] Although in FIG. 1, two loudspeakers 112, 120 are illustrated, ASRS 100 may include more or fewer loudspeakers. For example, ASRS 100 may have more than two loudspeakers for right, left, front and back balance of sound in vehicle cabin 126. As another example, ASRS 100 may include five loudspeakers (and a subwoofer) for 5.1 channel surround sound. It is understood that, in general, ASRS 100 may include one or more loudspeakers.

[0038] User interface 106 may include any suitable user interface capable of providing parameters for one or more of processor 114, indicator 116, VCRs 112, 120, VCM 118 and ambient microphones 104, 108, 110, 122. User interface 106 may include, for example, one or more buttons, a pointing device, a keyboard and/or a display device.

[0039] Processor 114 may also issue alerts to vehicle operator 124, for example, via indicator 116. Indicator 116 may provide alerts via a visual indication, an auditory indication (such as a tonal alert) and/or a haptic indication. Indicator 116 may include any suitable indicator such as (without being limited to): a display (such as a heads-up display), a loudspeaker or a haptic transducer (for example, mounted in the vehicle's steering wheel or operator seat)..

[0040] In an exemplary embodiment, processor 114 may also use ambient microphones 104, 108, 110, 122 and/or VCM 118 and VCRs 112, 120 to cancel a background noise component (such as road noise) in vehicle cabin 126. For example, the noise cancellation may be centered at the position of vehicle operator 124.

[0041] Ambient microphones 104, 108, 110, 122 may be positioned on vehicle 102 (for example, on an exterior of vehicle 102 or any other suitable location) such that ambient microphones 104, 108, 110, 122 may transduce sound that is external to vehicle 102. In general, ambient microphones 104, 108, 110, 122 may be configured to detect specific sounds in a vicinity of vehicle 102. Although four ambient microphones 104, 108, 110, 122 are illustrated in the positions (i.e., front, right, left and rear of vehicle 102) shown in FIG. 1, in general, system 100 may include any number of microphones and at least one ambient sound microphone. An ambient sound signal (from one or more of ambient microphones 104, 108, 110, 122) may also be mixed with AC signal 107 before being presented through at least one cabin loudspeaker 112, 120.

[0042] According to an exemplary embodiment, processor 114 may determine a sound pressure level (SPL) of vehicle cabin 126 (referred to herein as the cabin SPL) by analyzing a signal level and signal gain reproduced with at least one of loudspeakers 112, 120, and the sensitivity of respective loudspeakers 112, 120. In another exemplary embodiment, processor 114 may determine the cabin SPL via VCM 118. Use of VCM 118 may allow consideration of other sound sources in vehicle cabin 126 (i.e., other than sound sources contributed by loudspeakers 112, 120), such as an air conditioning system, and sound from other passengers in vehicle 102.

[0043] ASRS 100 may be coupled to a remote location (not shown), for example, by wireless communication. Information collected by ASRS 100 may be provided to the remote location (such as for further analysis).

[0044] Referring to FIG. 2, a block diagram of ASRS 100 in accordance with an exemplary embodiment is shown. As illustrated, the ASRS 100 can include processor 114 operatively coupled to the Ambient Sound Microphone (ASM) 201, one or more VCRs 112, 120, and VCM 118 via one or more Analog to Digital Converters (ADC) 202 and Digital to Analog Converters (DAC) 203. In FIG. 2, ASM 201 may represent one or more of ambient microphones 104, 108, 110, 122 shown in FIG. 1. ASRS 100 can include an audio interface 212 operatively coupled to the processor 114 to receive AC signal 107 (for example, from a media player, a cell phone, voice mail), and deliver the AC signal 107 to the processor 114.

[0045] The processor 114 may include sound signature detection block 214 and may monitor the ambient sound captured by the ASM 201 for warning sounds in the environment, such as an alarm (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.) , voice (e.g., "help", "stop", "police", etc.), or specific noise type (e.g., breaking glass, gunshot, etc.). The memory 208 can store sound signatures for previously learned warning sounds to which the processor 114 refers for detecting warning sounds. The sound signatures can be resident in the

memory 208 or downloaded to processor 114 via the transceiver 204 during operation as needed. Upon detecting a warning sound, the processor 114 can report the warning to the vehicle operator 124 (also referred to herein as user 124) via audio delivered from the VCRs 112, 120 to the vehicle cabin.

[0046] The processor 114 responsive to detecting warning sounds can adjust the audio content signal 107 and the warning sounds delivered to the vehicle cabin 126. The processor 114 can actively monitor the sound exposure level inside the vehicle cabin 126 and adjust the audio to within a safe and subjectively optimized listening level range. The processor 114 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or a digital signal processor (DSP) with associated storage memory 208 such as Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations.

[0047] The ASRS 100 can further include a transceiver 204 that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (Wi-Fi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver 204 can also provide support for dynamic downloading over-the-air to the ASRS 100. It should be noted also that next generation access technologies can also be applied to the present disclosure.

[0048] The power supply 210 can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the ASRS 100 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 210 to improve sensory input via haptic vibration (for example, via indicator 116 (FIG. 1) configured as a haptic indicator), e.g. connected to the vehicle steering wheel or vehicle operator chair. As an example, the processor 114 can direct the motor to vibrate or pulse responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0049] FIG. 3 is a flowchart of a method 300 for vehicle ambient sound monitoring and warning detection in accordance with an exemplary embodiment. The method 300 can be practiced with more or fewer than the number of steps shown and is not limited to the order shown. To describe the method 300, reference will be made to components of FIG 2 , although it is understood that the method 300 can be implemented in any other manner using other suitable components.

[0050] As shown in step 302, the processor 114 can monitor the environment for warning sounds, such as an alarm, a horn, a voice, or a noise. Each of the warning sounds can have certain identifiable features that characterize the sound. The features

can be collectively referred to as a sound signature which can be used for recognizing the warning sound. As an example, the sound signature may include statistical properties or parametric properties of the warning sound. For example, a sound signature can describe prominent frequencies with associated amplitude and phase information. As another example, the sound signature can contain principal components identifying the most likely recognizable features of a warning sound.

[0051] At step 304, the processor 114 may detect the warning sounds within the environment based on the sound signatures. As described below, feature extraction techniques may be applied to the ambient sound captured at the ASM 201 to generate the sound signatures. Pattern recognition approaches may be applied based on known sound signatures to detect the warning sounds from their corresponding sound signatures. More specifically, sound signatures may be compared to learned models to identify a corresponding warning sound.

[0052] At step 306, the processor 114 may adjust sound delivered to the vehicle cabin 126 in view of a detected warning sound. Upon detecting a warning sound in the ambient sound of the user's environment, the processor 114, at step 308, may generate an audible alarm within the vehicle cabin 126 that identifies the detected sound signature.

[0053] The audible alarm can be a reproduction of the warning sound, an amplification of the warning sound (or the entire ambient sound), a text-to-speech message (e.g., synthetic voice) identifying the warning sound, a haptic vibration via indicator 116 (configured as a haptic indicator), or an audio clip. For example, the processor 114 can generate a sound bite (i.e., audio clip) corresponding to the detected warning sound such as an ambulance, fire engine, or other environmental sound. As another example, the processor 114 can synthesize a voice to describe the detected warning sound (e.g., "ambulance approaching"). At step 310, processor 114 may send a message to a mobile device identifying the detected sound signature (e.g., "alarm sounding").

[0054] FIG. 4 illustrates system modes of ASRS 100 in accordance with an exemplary embodiment. The system mode may be manually selected by user 124, for example, by pressing a button; or automatically selected, for example, when the processor 114 detects it is in an active listen state or in a media state. As shown in FIG. 4, the system mode can correspond to Signature Sound Pass Through Mode (SSPTM), Signature Sound Boost Mode (SSBM), Signature Sound Rejection Mode (SSRJM), Signature Sound Attenuation Mode (SSAM), and Signature Sound Replacement Mode (SSRM).

[0055] In SSPTM mode, ambient sound captured at the ASM 201 is passed transparently to the VCRs 112, 120 for reproduction within the vehicle cabin 126. In this

mode, the sound produced in the vehicle cabin 126 sufficiently matches the ambient sound outside the vehicle cabin 126, thereby providing a "transparency" effect. That is, the loudspeakers 112, 120 in the vehicle cabin 126 recreate the sound captured at the ASM 201. The processor 114, by way of sound measured at the VCM 118, may adjust the properties of sound delivered to the vehicle cabin 126 so the sound within the occluded vehicle cabin 126 is the same as the ambient sound outside the vehicle 102.

[0056] In SSBM mode, warning sounds and/or ambient sounds are amplified upon the processor 114 detecting a warning sound. The warning sound can be amplified relative to the normal level received, or amplified above an audio content level if audio content is being delivered to the vehicle cabin 126.

[0057] In SSRJM mode, sounds other than warning sounds may be rejected upon the processor 114 detecting a specific sound signature. The specific sound can be minimized relative to the normal level received. In SSAM mode, sounds other than warning sounds can be attenuated. For example, in both SSRJM and SSAM modes, annoying sounds or noises not associated with warning sounds can be suppressed. For instance, by way of a learning session, the user 124 can establish which sounds are considered warning sounds (e.g., "ambulance") and which sounds are considered non-warning sounds (e.g. "jackhammer"). The processor 114 upon detecting non-warning sounds can thus attenuate or reject these sounds within the vehicle cabin 126.

[0058] In SSRM mode, warning sounds detected in the environment can be replaced with audible warning messages. For example, the processor 114, upon detecting a warning sound, can generate synthetic speech identifying the warning sound (e.g., "ambulance detected"). In such regard, the processor 114 may audibly report the warning sound identified, thereby relieving the user 124 from having to interpret the warning sound. The synthetic speech can be mixed with the ambient sound (e.g., amplified, attenuated, cropped, etc.), or played alone with the ambient sound muted.

[0059] FIG. 5 is a flowchart of a method 500 for sound signature detection in accordance with an exemplary embodiment. The method 500 can be practiced with more or fewer than the number of steps shown and is not limited to the order shown. To describe the method 500, reference will be made to components of FIG 2 , although it is understood that the method 500 can be implemented in any other manner using other suitable components.

[0060] The method can start at step 502, in which the processor 114 can enter a learn mode. Notably, the processor 114 upon completion of a learning mode or previous learning configuration can start instead at step 520. In the learning mode of step 502, the processor 114 can actively generate and learn sound signatures from ambient sounds within the environment. In learning mode, the processor 114 can also receive

previously trained learning models to use for detecting warning sounds in the environment.

[0061] In an active learning mode, the user 124 can press a button or otherwise (e.g. voice recognition) initiate a recording of ambient sounds in the environment. For example, the user can, upon hearing a new warning sound in the environment ("car horn"), activate the processor 114 to learn the new warning sound. Upon generating a sound signature for the new warning sound, it can be stored in the user defined database 504. In another arrangement, the processor 114, upon detecting a unique sound, characteristic to a warning sound, can ask the user 124 if they desire to have the sound signature for the unique sound learned. In such regard, the processor 114 may actively sense sounds and may query the user 124 about the environment to learn the sounds. Moreover, the processor 114 can organize learned sounds based on environmental context, for example, in city and country environments.

[0062] In an exemplary embodiment, ASRS 100 (FIG. 2) may provide for delayed recording, to allow a previously encountered sound to be learned. For example, ASRS 100 may include a buffer to store ambient sounds recorded for a period of time. User 124 may review the recorded ambient sounds and select one or more of these recorded ambient sounds for learning (such as via user interface 106 (FIG. 1)).

[0063] In another learning mode, trained models can be retrieved from an on-line database 506 for use in detecting warning sounds. The previously learned models can be transmitted on a scheduled basis to the processor 114, or as needed, depending on the environmental context. For example, upon the processor 114, upon detecting traffic noise, may retrieve sound signature models associated with warning sounds (e.g., ambulance, police car) in traffic. In another embodiment, upon the processor 114 detecting conversational noise (e.g. people talking), sound signature models for verbal warnings ("help", "police") may be retrieved. Groups of sound signature models may be retrieved based on the environmental context or on user directed action.

[0064] As shown in step 508, the ASRS processor 114 can also generate speech recognition models for warning sounds corresponding to voice, such as "help", "police", "fire", etc. The speech recognition models may be retrieved from the on-line database 506 or the user defined database 504. In the latter for example, the user 124 may say a word or enter a text version of a word to associate the word with a verbal warning sound. For instance, the user 124 may define a set of words of interest along with mappings to their meanings, and then use keyword spotting to detect their occurrences. If the user 124 enters an environment wherein another individual says the same word (e.g., "help") the processor 114 may inform the user 124 of the verbal warning sound.

[0065] For other acoustic sounds, the processor 114 may generate sound signature models as shown in step 510. Notably, the processor 114 itself may generate the sound signature models, or transmit the captured warning sounds to external systems (e.g., a remote server) that generate the sound signature models. Such learning may be conducted off-line in a training phase, and the processor 114 can be uploaded with the new learning models.

[0066] It should also be noted that the learning models can be updated during use of the ASRS 100, for example, when the processor 114 detects warning sounds. The detected warning sounds can be used to adapt the learning models as new warning sound variants are encountered. For example, the processor 114 upon detecting a warning sound, can use the sound signature of the warning sound to update the learned models in accordance with the training phase. In such an embodiment a first learned model is adapted based on new training data collected in the environment by the processor 114. In such regard, for example, a new set of "horn" warning sounds could be included in real-time training without discarding the other "horn" sounds already captured in the existing model.

[0067] Upon completion of learning, uploading, or retrieval of sound signature models, the processor 114 can monitor and report warning sounds within the environment. As shown in step 520, ambient sounds (e.g. an input signal) within the environment are captured by the ASM 201. The ambient sounds can be digitized by way of the ADC 202 and stored temporarily to a data buffer in memory 208 as shown in step 522. The data buffer may be capable of holding enough data to allow for generation of a sound signature as will be described ahead in FIG. 7.

[0068] In another configuration, the processor 114 can implement a "look ahead" analysis system by way of the data buffer for reproduction of pre-recorded audio content, using a data buffer to offset the reproduction of the audio signal. The look-ahead system allows the processor 114 to analyze potentially harmful audio artifacts (e.g. high level onsets, bursts, etc.) either received from an external media device, or detected with the ambient microphones 201, in-situ before it is reproduced. The processor 114 can thus mitigate the audio artifacts in advance to reduce timbral distortion effects caused by, for instance, attenuating high level transients.

[0069] At step 524, signal conditioning techniques may be applied to the ambient sound, for example, to suppress noise or gate the noise to a predetermined threshold. Other signal processing steps such as threshold detection shown in step 526 may be used to determine whether ambient sounds should be evaluated for warning sounds. For instance, to conserve computational processing resources (e.g., battery, processor), only ambient sounds that exceed a predetermined power level may be evaluated for warning

sounds. Other metrics such as signal spectrum, duration, and stationarity may be considered in determining whether the ambient sound is analyzed for warning sounds. Notably, other metrics (e.g., context aware) may also be used to determine when the ambient sound should be processed for warning sound detection.

[0070] If at least one property (e.g., power, spectral shape, duration, etc.) of the ambient sound exceeds a threshold (or adaptive threshold), the processor 114 at step 530 can proceed to generate a sound signature for the ambient sound. In one embodiment the sound signature is a feature vector which can include statistical parameters or salient features of the ambient sound.

[0071] An ambient sound with a warning sound (e.g. "bell", "siren"), such as shown in step 532, is generally expected to exhibit features similar to sound signatures for similar warning sounds (e.g. "bell", "siren") stored in the user defined database 504 or the on-line database 506. The processor 114 can also identify a direction and speed of the sound source if it is moving, for example, by evaluating Doppler shift as shown in step 534 and 536. The processor 114, by way of beam-forming among multiple ASMs 201 may also estimate a direction of a sound source generating the warning sound.

[0072] The speed and bearing of the sound source can also be estimated using pitch analysis to detect changes predicted by Doppler effect, or alternatively by an analysis in changes in relative phase and magnitude between the two ASM signals. The processor 114, by way of a sound recognition engine, may detect general warning signals such as car horns or emergency sirens (and other signals referenced by ISO 7731) using spectral and temporal analysis.

[0073] The processor 114 can also analyze the ambient sound to determine if a verbal warning (e.g. "help", "police", "excuse me") is present. As shown in step 540, the sound signature of the ambient sound can be analyzed for speech content. For example, the sound signature may be analyzed for voice information, such as vocal cord pitch periodicities, time-varying voice formant envelopes, or other articulation parameter attributes. Upon detecting the presence of voice in the ambient sound, the processor 114 can perform key word detection (e.g. "help") in the spoken content as shown in step 542. Speech recognition models as well as language models may be used to identify key words in the spoken content. As previously noted, the user 124 may say or enter in one or more warning sounds that may be mapped to associated learning models for sound signature detection.

[0074] As shown in step 552, the user 124 may also provide user input to direct operation of the processor 114, for example, to select an operational mode as shown in 550. As one example, the operational mode can enable, disable or adjust monitoring of warning sounds. For example, in a listening mode, the processor 114 may mix audio

content with ambient sound while monitoring for warning sounds. In a quiet mode, the processor 114 may suppress or attempt to actively cancel all noises except detected warning sounds.

[0075] The user input may be in the form of a physical interaction (e.g., button press) or a vocalization (e.g., spoken command). The operating mode can also be controlled by a prioritizing module as shown in step 554. The prioritizing module may prioritize warning sounds based on severity and context. For example, if the user 124 is in a phone call, and a warning sound is detected, the processor 114 may audibly inform the user 124 of the warning and/or present a text message of the warning sound. If the user 124 is listening to music or a voice communication, and a warning sound is detected, the processor 114 may automatically shut off the music or voice audio and alert the user. The user 124, by way of user interface 106 (FIG. 1) or an administrator, may rank warning sounds and instruct the processor 114 how to respond to warnings in various contexts.

[0076] FIG. 6 is a flowchart of a method 600 for managing audio delivery based on detected sound signatures in accordance with an exemplary embodiment. The method 600 can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method 600, reference will be made to components of FIG 2 , although it is understood that the method 600 can be implemented in any other manner using other suitable components.

[0077] As noted previously, the audio interface 212 can supply audio content (e.g., music, cell phone, voice mail, etc.) to the processor 114. In such regard, the user 124 may listen to music, talk on the phone, receive voice mail, or perform other audio related tasks while the processor 114 additionally monitors warning sounds in the environment. During normal use, when a warning sound is not present, the processor 114 may operate normally to recreate the sound experience requested by the user 124. If however the processor 114 detects a warning sound, the processor 114 may manage audio content delivery to notify the user 124 of the warning sound. Managing audio content delivery can include adjusting or overriding other current audio settings.

[0078] By way of example, as shown in step 602, the audio interface 212 receives audio content from a media player, such as a portable music player, or cell phone. The audio content can be delivered to the user's vehicle cabin by way of the VCRs 112, 120 as shown in step 604.

[0079] At step 606, the processor 114 monitors ambient sound in the environment captured at the ASM 201. Ambient sound may be sampled at sufficiently data rates (e.g. 8 kHz, 16 kHz, and 32 KHz) to allow for feature extraction of sound signatures. Moreover, the processor 114 may adjust the sampling rate based on the information

content of the ambient signal. For example, upon the ambient sound exceeding a first threshold, the sampling rate may be set to a first rate (e.g., 4 KHz). As the ambient sound increases in volume, or as prominent features are identified, the sampling rate may be increased to a second rate (e.g., 8 KHz) to increase signal resolution. Although, the higher sampling rate may improve resolution of features, the lower sampling rate may preserve use of computational resources for minimally sufficient feature resolution (e.g., power supply 210, processor 114).

[0080] If at step 608, a sound signature is detected, the processor 114 may determine a priority of the detected sound signature (at step 610). The priority establishes how the processor 114 manages audio content. Notably, warning sounds for various environmental conditions and user experiences can be learned. Accordingly, the user 124 or an administrator, can establish priorities for warning sounds. Moreover, these priorities may be based on environmental context. For example, if a user 124 is in a warehouse where loading vehicles emit a beeping sound, sound signatures for such vehicles can be given the highest priority. A user 124 may also prioritize learned warning sounds, for example, via a user interface on a paired device (e.g., cell phone), or via speech recognition (e.g., "prioritize – 'ambulance' – high").

[0081] Upon detecting a warning sound and identifying a priority, the processor 114, at step 612, selectively manages at least a portion of the audio content based on the priority. For example, if the user 124 is listening to music during the time a warning sound is detected, the processor 114 may decrease the music volume to present an audible notification. This may represent one indication that the processor 114 has detected a warning sound.

[0082] At step 614, the processor 114 may further present an audible notification to the user 124. For example, upon detecting a "horn" sound, a speech-to-text message can be presented to the user 124 to audibly inform them that a horn sound has been detected (e.g., "horn detected"). Information related to the warning sound (e.g., direction, speed, priority, etc.) may also be presented with the audible notification.

[0083] In a further arrangement, the processor 114 may send a message to a device operated by the user 124 to visually display the notification, as shown in step 616. For example, if the user has disengaged audible notification, the processor 114 may transmit a text message to a paired device (e.g., a cell phone) containing the audible warning (or to indicator 116 configured as a visual indicator). Moreover, the processor 114 may beacon out an audible alarm to other devices within a vicinity, for example via Wi-Fi (e.g., IEEE 802.16x). Other devices in a proximity of the user 124 may sign up to receive audible alarms from the processor 114. In such regard, the processor 114 can

beacon a warning notification to other devices in the area to share warning information with other people.

[0084] FIG. 7 is a flowchart of a method 700 further describing sound signature detection in accordance with an exemplary embodiment. The method 700 can be practiced with more or fewer than the number of steps shown and is not limited to the order shown. The method 700 can begin in a state in which the processor 114 is actively monitoring warning sounds in the environment.

[0085] At step 711, ambient sound captured from the ASM 201 may be buffered into short term memory as frames. As an example, the ambient sound may be sampled at 8 KHz with 10-20 ms frame sizes (80 to 160 samples). The frame size may also vary depending on the energy level of the ambient sound. For example, the processor 114 upon detecting low level sounds (e.g., about 70-74 dB SPL) may use a frame size of about 30 ms, and update the frame size to about 10 ms as the power level increases (e.g., greater than about 86 dB SPL). The processor 114 may also increase the sampling rate in accordance with the power level and/or a duration of the ambient sound. (A longer frame size with lower sampling may compromise resolution for computational resources.) The data buffer is desirably of sufficient length to hold a history of frames (e.g., about 10-15 frames) for short-term historical analysis.

[0086] At step 712, the processor 114 may perform feature extraction on the frame as the ambient sound is buffered into the data buffer. As one example, feature extraction may include performing a filter-bank analysis and summing frequencies in auditory bandwidths. Features may also include Fast Fourier Transform (FFT) coefficients, Discrete Cosine Transform (DCT) coefficients, cepstral coefficients, partial autocorrelation (PARCOR) coefficients, wavelet coefficients, statistical values (e.g., energy, mean, skew, variance), parametric features, or any other suitable data compression feature set.

[0087] Additionally, dynamic features, such as derivatives of any order, may be added to the static feature set. As one example, mel-frequency-cepstral analysis may be performed on the frame to generate between about 10-16 mel-frequency-cepstral coefficients. The small number of coefficients represent features that may be compactly stored to memory for that particular frame. Such front end feature extraction techniques may reduce the amount of data used to represent the data frame.

[0088] At step 713, the features may be incorporated as a sound signature and compared to learned models, for example, those retrieved from the warning sounds database 718 (e.g., user defined database 504 or the on-line database 506 of FIG. 5). A sound signature may be defined as a sound in the user's ambient environment which has significant perceptual saliency. As an example, a sound signature may correspond to an

alarm, an ambulance, a siren, a horn, a police car, a bus, a bell, a gunshot, a window breaking, or any other warning sound, including voice. The sound signature may include features characteristic to the sound. As an example, the sound signature may be classified by statistical features of the sound (e.g., envelope, harmonics, spectral peaks, modulation, etc.).

[0089] Notably, each learned model used to identify a sound signature has a set of features specific to a warning sound. For example, a feature vector of a learned model for an "alarm" is sufficiently different from a feature vector of a learned model for a "bell sound". Moreover, the learned model may describe interconnectivity (e.g., state transitions, emission probabilities, initial probabilities, synaptic connections, hidden layers) among the feature vectors (e.g. frames). For example, the features of a "bell" sound may change in a specific manner compared to the features of an "alarm" sound. The learned model may be a statistical model such as a Gaussian mixture model (GMM), a Hidden Markov Model (HMM), a Bayes Classifier, or a Neural Network (NN) that requires training.

[0090] In the foregoing, a Gaussian Mixture Model (GMM) is presented, although it should be noted that any of the above models may be used for sound signature detection. In this case, each warning sound may have an associated GMM used for detecting the warning sound. As an example, the warning sound for an "alarm" may have its own GMM, and a warning sound for a "bell" may have its own GMM. Separate GMMs may also be used as a basis for the absence of the sounds ("anti-models"), such as "not alarm" or "not bell." Each GMM may provide a model for the distribution of the feature statistics for each warning sound in a multi-dimensional space.

[0091] Upon presentation of a new feature vector, the likelihood of the presence of each warning sound may then be calculated. In order to detect a warning sound, each warning sound's GMM may be evaluated relative to its anti-model, and a score related to the likelihood of that warning sound may be computed. A threshold may be applied directly to this score to decide whether the warning sound is present or absent. Similarly, the sequence of scores may be relayed to yet another module which uses a more complex rule to decide presence or absence. Examples of such rules include linear smoothing or median filtering.

[0092] As previously noted, a HMM model or NN model with their associated connection logic may be used in place of each GMM for each learning model. For example, each warning sound in the database 718 may have a corresponding HMM. A sound signature for a warning sound captured at the ASM 201 in ambient sound may be processed through a lattice network (e.g. Viterbi network) for comparison to each HMM, to determine which HMM, if any, corresponds to the warning sound. Alternatively, in a

trained NN, the sound signature may be input to the NN, where the output states of the NN correspond to warning sound indices. The NN may include various topologies such as a Feed-Forward, Radial Basis Function, Hopfield, Time-Delay Recurrent, or other optimized topologies for real-time sound signature detection.

[0093] At step 714, a distortion metric is performed with each learned model to determine which learned models are closest to the captured feature vector (e.g., sound signature). The learned model with the smallest distortion (e.g., mathematical distance) is generally considered the correct match, or recognition result. It should also be noted that the distortion may be calculated as part of the model comparison in step 713. This is because the distortion metric may depend on the type of model used (e.g., HMM, NN, GMM, etc.) and in fact may be internal to the model (e.g. Viterbi decoding, back-propagation error update, etc.). The distortion module is merely presented in FIG. 7 as a separate component to suggest use with other types of pattern recognition methods or learning models.

[0094] Upon evaluating the feature vector (e.g. sound signature) against the candidate warning sound learned models, the ambient sound at step 715 may be classified as a warning sound. Each of the learned models may be associated with a score. For example, upon the presentation of a sound signature, each GMM may produce a score. The scores may be evaluated against a threshold, and the GMM with the highest score may be identified as the detected warning sound. For example, if the learned model for the "alarm" sound produces the highest score (e.g., smallest distortion result) compared to other learned models, the ambient sound may be classified as an "alarm" warning sound.

[0095] The classification step 715 also takes into account likelihoods (e.g. recognition probabilities). For example, as part of the step of comparing the sound signature of the unknown ambient sound against all the GMMs for the learned models, each GMM may produce a likelihood result, or output. As an example, these likelihood results may be evaluated against each other or in a logical context, to determine the GMM considered "most likely" to match the sound signature of the warning sound. The processor 114 may then select the GMM with the highest likelihood or score via soft decisions.

[0096] The processor 114 may continually monitor the environment for warning sounds, or monitor the environment on a scheduled basis. In one arrangement, the processor 114 may increase monitoring in the presence of high ambient noise possibly signifying environmental danger or activity. Upon classifying an ambient sound as a warning sound the processor 114, at step 716, the ASRS 100 may generate an alarm. As previously noted, the processor 114 may mix the warning sound with audio content, amplify the warning sound, reproduce the warning sound, and/or deliver an audible

message. As one example, spectral bands of the audio content that mask the warning sound may be suppressed to increase an audibility of the warning sound. This serves to notify the user 124 of a warning sounded detected in the environment, of which the user 124 may not be aware of depending on their environmental context.

[0097] As an example, the processor 114 may present an amplified audible notification to the user via the VCRs 112, 120. The audible notification may be a synthetic voice identifying the warning sound (e.g. "car alarm"), a location or direction of the sound source generating the warning sound (e.g. "to your left"), a duration of the warning sound (e.g., "3 minutes") from initial capture, and any other information (e.g., proximity, severity level, etc.) related to the warning sound. Moreover, the processor 114 may selectively mix the warning sound with the audio content based on a predetermined threshold level. For example, the user 124 may prioritize warning sound types for receiving various levels of notification, and/or identify the sound types as desirable of undesirable.

[0098] FIG. 8, presents a pictorial diagram 800 for mixing ambient sounds and warning sounds with audio content. In the illustration show, the processor 114 is directing music 136 to the vehicle cabin 126 via VCR 120 while simultaneously monitoring warning sounds in the environment. At time, T, the processor 114, upon detecting a warning sound (signature 135), can lower the music volume from the media player 150 (graph 141), and increase the volume of the ambient sound received at the ASM 201 (graph 142). Other mixing arrangements are herein contemplated.

[0099] In such regard, there is a smooth audio transition between the music 136 and the warning sound 135. Notably, the ramp up and down times can also be adjusted based on the priority of the warning sound. For example, in an extreme case, the processor 114 may immediately shut off the music, and present the audible warning. Other various implementations for mixing audio and managing audio content delivery are herein contemplated.

[00100] Moreover, the audio content may be managed with other media devices (e.g., a cell phone). For example, upon detecting a warning sound, the processor 114 may inform the user 124 and the called party of a warning sound. In such regard, the user 124 does not need to inform the called party since they also receive the notification, which may save them time to explain an emergency situation.

[00101] As one example, the processor 114 may spectrally enhance the audio content in view of the ambient sound. Moreover, a timbral balance of the audio content may be maintained by taking into account level dependent equal loudness curves and other psychoacoustic criteria (e.g., masking) associated with a personalized hearing level (PHL) 430. For example, auditory queues in a received audio content may be enhanced

based on the PHL 430 and a spectrum of the ambient sound captured at the ASM 201. Frequency peaks within the audio content may be elevated relative to ambient noise frequency levels and in accordance with the PHL 430 to permit sufficient audibility of the ambient sound. The PHL 430 reveals frequency dynamic ranges that may be used to limit the compression range of the peak elevation in view of the ambient noise spectrum.

[00102] In one arrangement, the processor 114 may compensate for a masking of the ambient sound by the audio content. Notably, the audio content, if sufficiently loud, may mask auditory queues in the ambient sound, which can: i) potentially cause hearing damage, and ii) prevent the user 124 from hearing warning sounds in the environment (e.g., an approaching ambulance, an alarm, etc.). Accordingly, the processor 114 may accentuate and attenuate frequencies of the audio content and ambient sound to permit maximal sound reproduction while simultaneously permitting audibility of ambient sounds.

[00103] In one exemplary embodiment, the processor 114 may narrow noise frequency bands within the ambient sound to permit sensitivity to audio content between the frequency bands. The processor 114 may also determine if the ambient sound contains salient information (e.g., warning sounds) that should be un-masked with respect to the audio content. If the ambient sound is not relevant, the processor 114 may mask the ambient sound (e.g., increase levels) with the audio content until warning sounds are detected.

[00104] FIG. 9 is a flowchart of a method 900 for updating the sound signature detection library dependent on the vehicle location. At step, 902, a current vehicle location may be acquired. The location of vehicle 102 may be determined with a number of methods, including: a Global Positioning System (GPS) for determining a GPS location and cell-phone signal cell codes for triangulating the location (for example, based on a signal strength of the received signal(s) from cell phone networks).

[00105] The acquired vehicle location may be used, at step 904, to determine which "sound library cell" the vehicle 102 is located in. The sound library cell may refer to a geographic region, such as a country, state, province or city. The sound library cell may contain the target sound signatures which the vehicle's sound recognition system detects (for example, for a particular sound signature of a particular fire engine siren, there may be associated GMM statistics, as described above). The cell library may be determined from the acquired location (for example, using a look-up table), or may be acquired directly from coded signals embedded in the received location signal (for example, a cell-phone signal or a radio signal).

[00106] If, at step 906, the vehicle library cell location is determined to have changed, then the current sound signature library of the vehicle is updated (at step 910). For

example, the current vehicle sound signature library may be updated from data provided by an on-line sound signature library 908, where the new sound signature data may be received wirelessly or from a storage device located on the vehicle 102.

[00107] Although the invention has been described in terms of automotive sound recognition systems and methods for enhancing situation awareness in a vehicle, it is contemplated that one or more steps and/or components may be implemented in software for use with microprocessors/general purpose computers (not shown). In this embodiment, one or more of the functions of the various components and/or steps described above may be implemented in software that controls a computer. The software may be embodied in non-transitory tangible computer readable media (such as, by way of non-limiting example, a magnetic disk, optical disk, flash memory, hard drive, etc.) for execution by the computer.

[00108] Although the invention is illustrated and described herein with reference to specific embodiments, the invention is not intended to be limited to the details shown. Rather, various modifications may be made in the details within the scope and range of equivalents of the claims and without departing from the invention.

What is Claimed:

1. A sound recognition system for a vehicle comprising:

at least one ambient sound microphone (ASM), disposed on the vehicle, configured to capture ambient sound external to the vehicle;

at least one vehicle cabin receiver (VCR) configured to deliver audio content to a vehicle cabin of the vehicle; and

a processor, coupled to the at least one ASM and the at least one VCR, the processor configured to detect at least one sound signature in the ambient sound and to adjust the audio content delivered to the vehicle cabin based on the detected at least one sound signature.

2. The system according to claim 1, wherein the sound signature includes at least one of a non-verbal warning sound or a verbal warning sound.

3. The system according to claim 2, wherein the non-verbal warning sound includes at least one of an alarm, a horn, a siren or a noise.

4. The system according to claim 2, wherein the verbal warning sound includes one or more spoken words associated with a verbal warning.

5. The system according to claim 1, wherein the processor is configured to selectively adjust a volume of the audio content delivered to the vehicle cabin when the at least one sound signature is detected.

6. The system according to claim 1, wherein the processor is configured to reproduce the ambient sound associated with the at least one sound signature within the vehicle cabin.

7. The system according to claim 1, wherein the processor is configured to selectively mix the audio content with the ambient sound when the at least one sound signature is detected.

8. The system according to claim 1, further including a memory configured to store a target sound captured by the at least one ASM for learning the corresponding sound signature.

9. The system according to claim 8, wherein the memory is configured to store the target sound responsive to an indication by a user of the system.

10. The system according to claim 8, wherein the memory is configured to store the target sound automatically by the system.

11. The system according to claim 1, further including an audio interface coupled to the processor to receive the audio content from at least one of a media player or a mobile phone.

12. The system according to claim 1, wherein the system is configured to transmit one or more of the ambient sound and the at least one sound signature to a remote location.

13. A method for increasing auditory situation awareness in a vehicle, the method comprising the steps of:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle;

monitoring the ambient sound for a target sound by detecting a sound signature corresponding to the target sound in the ambient sound; and

adjusting a delivery of audio content by at least one vehicle cabin receiver (VCR) to a vehicle cabin of the vehicle based on the target sound.

14. The method according to claim 13, wherein the adjusting of the delivery of the audio content includes mixing the target sound with the audio content for delivery to the vehicle cabin.

15. The method according to claim 14, wherein the target sound is mixed with the audio content in accordance with a priority of the target sound.

16. The method according to claim 13, wherein the adjusting of the delivery of the audio content includes at least one of:

passing the target sound to the at least one VCR,

amplifying the target sound for delivery to the vehicle cabin,

attenuating the target sound for delivery to the vehicle cabin,

generating an audible message based on the target sound for delivery to the vehicle cabin, or

replacing the target sound with a predetermined sound corresponding to the target sound for delivery to the vehicle cabin.

17. The method according to claim 13, the method further including:

detecting at least one of a direction of a sound source or a speed of the sound source generating the target sound from the sound signature; and

indicating the at least one of the direction or the speed of the sound source in the vehicle cabin.

18. The method according to claim 13, the method further including transmitting a warning notification to other devices.

19. The method of claim 13, wherein the target sound includes at least one of an alarm, a horn, a siren, a spoken utterance or a noise.

20. The method according to claim 13, wherein the detecting of the sound signature includes detecting a spoken utterance in the ambient sound associated with a verbal warning, the method further including:

indicating the verbal warning in the vehicle cabin.

21. The method according to claim 13, the method further including:

acquiring a current location of the vehicle;

associating the current location with a sound signature; and

updating a sound signature library containing a plurality of predetermined target sounds with the sound signature associated with the current location.

22. A method for sound signature detection for a vehicle, the method comprising:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle; and

receiving a directive to learn a sound signature within the ambient sound,

wherein a voice command or an indication from a user is received and is used to initiate the steps of capturing and learning.

23. The method according to claim 22, further including saving the sound signature at least one of locally on the vehicle or remotely to a server.

24. The method according to claim 22, further including adapting a previously learned warning sound model using the sound signature within the ambient sound.

25. A method for personalized listening in a vehicle, the method comprising:

capturing ambient sound external to the vehicle from at least one ambient sound microphone (ASM) disposed on the vehicle;

detecting a sound signature within the ambient sound that is associated with a warning sound; and

mixing the warning sound with audio content delivered to the vehicle cabin via at least one vehicle cabin receiver (VCR) in accordance with a priority of the warning sound and a personalized hearing level (PHL).

26. The method according to claim 25, wherein the detecting of the sound signature includes:

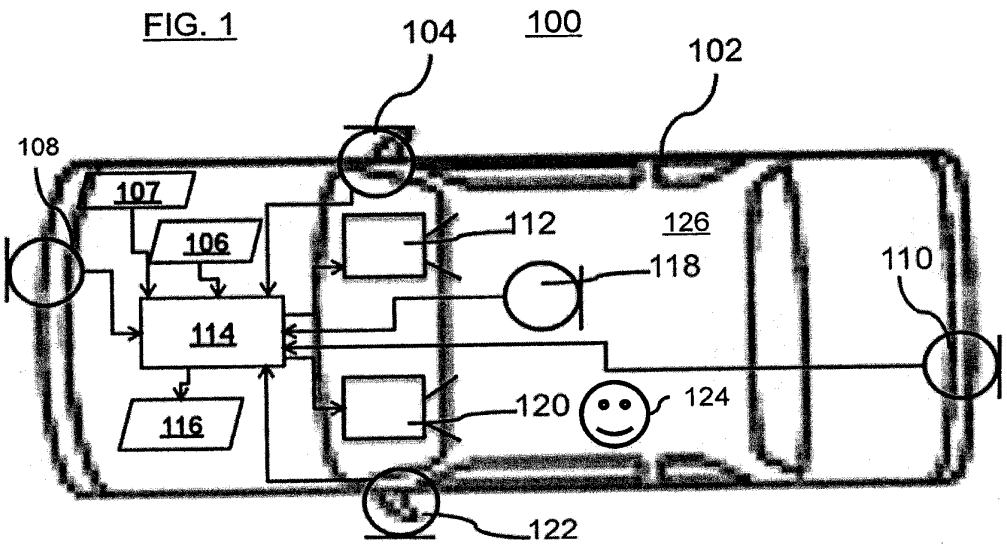
retrieving learned models from a database;

comparing the sound signature to the learned models; and

identifying the warning sound from the learned models responsive to the comparison.

27. The method according to claim 25, further including enhancing auditory queues in the warning sound relative to the audio content based on a spectrum of the ambient sound captured at the at least one ASM.

FIG. 1


104
100
102
108
107
106
112
126
118
110
114
116
120
124
122

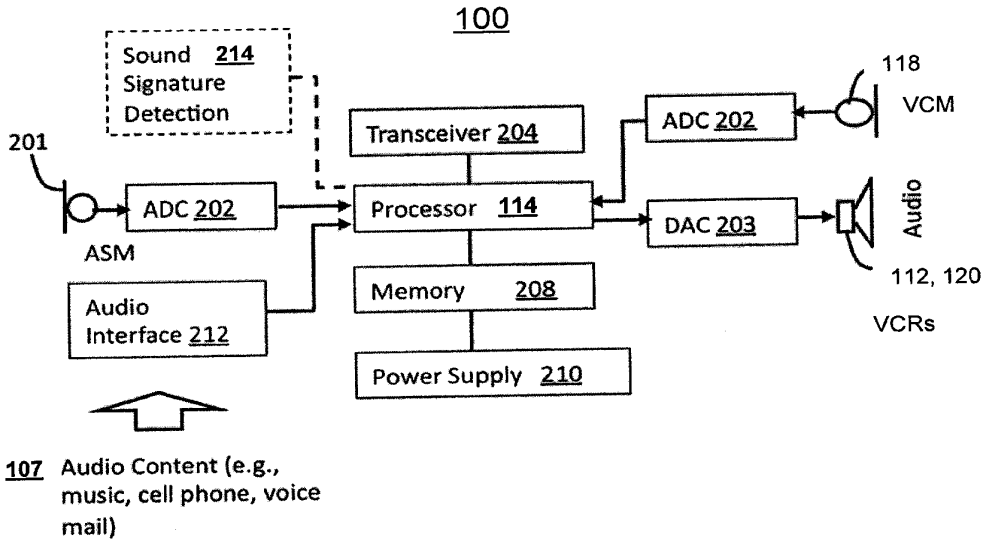

100
Sound 214
Signature
Detection
Transceiver 204
ADC 202
118
VCM
201
ADC 202
Processor 114
DAC 203
Audio
ASM
112, 120
Memory 208
Audio
Interface 212
VCRs
Power Supply 210
107 Audio Content (e.g., music, cell phone, voice mail)

FIG. 2

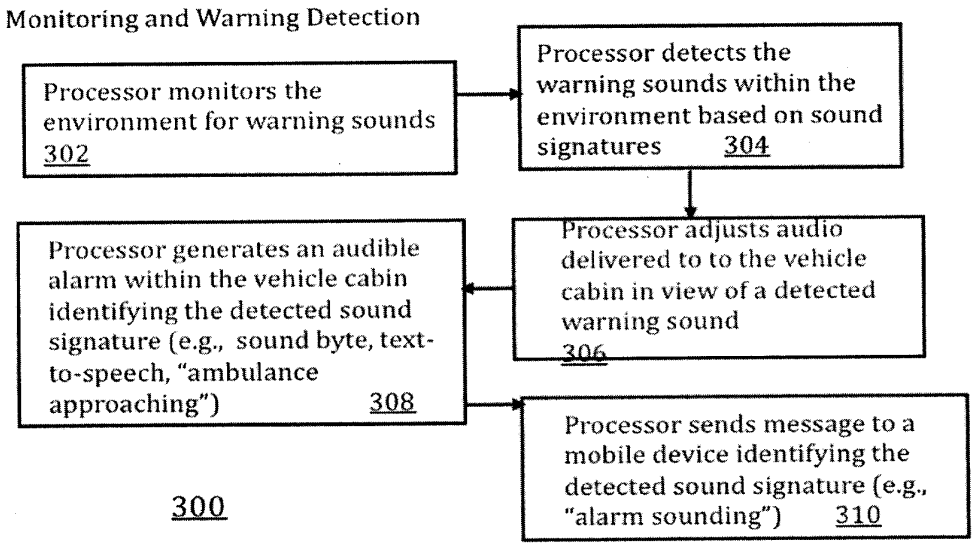

Monitoring and Warning Detection
Processor monitors the environment for warning sounds 302
Processor detects the warning sounds within the environment based on sound signatures 304
Processor adjusts audio delivered to to the vehicle cabin in view of a detected warning sound 306
Processor generates an audible alarm within the vehicle cabin identifying the detected sound signature (e.g., sound byte, text-to-speech, "ambulance approaching") 308
Processor sends message to a mobile device identifying the detected sound signature (e.g., "alarm sounding") 310
300

FIG. 3

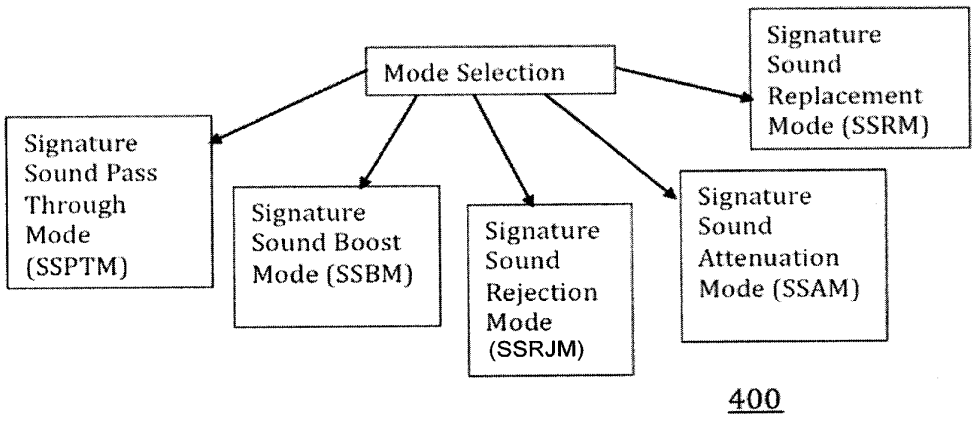

Mode Selection
Signature Sound Pass Through Mode (SSPTM)
Signature Sound Boost Mode (SSBM)
Signature Sound Rejection Mode (SSRJM)
Signature Sound Attenuation Mode (SSAM)
Signature Sound Replacement Mode (SSRM)
400

FIG. 4

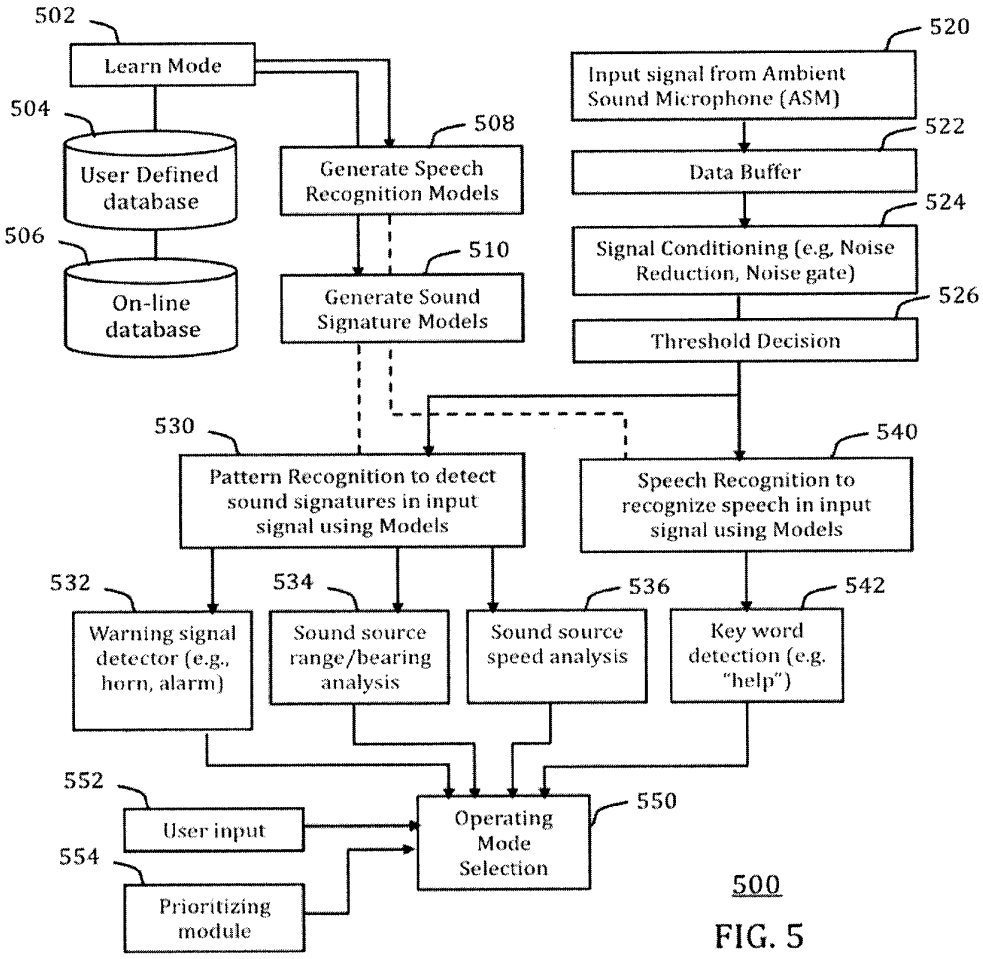

502
Learn Mode
504
User Defined database
506
On-line database
508
Generate Speech Recognition Models
510
Generate Sound Signature Models
520
Input signal from Ambient Sound Microphone (ASM)
522
Data Buffer
524
Signal Conditioning (e.g. Noise Reduction, Noise gate)
526
Threshold Decision
530
Pattern Recognition to detect sound signatures in input signal using Models
540
Speech Recognition to recognize speech in input signal using Models
532
Warning signal detector (e.g., horn, alarm)
534
Sound source range/bearing analysis
536
Sound source speed analysis
542
Key word detection (e.g. "help")
552
User input
554
Prioritizing module
550
Operating Mode Selection
500

FIG. 5

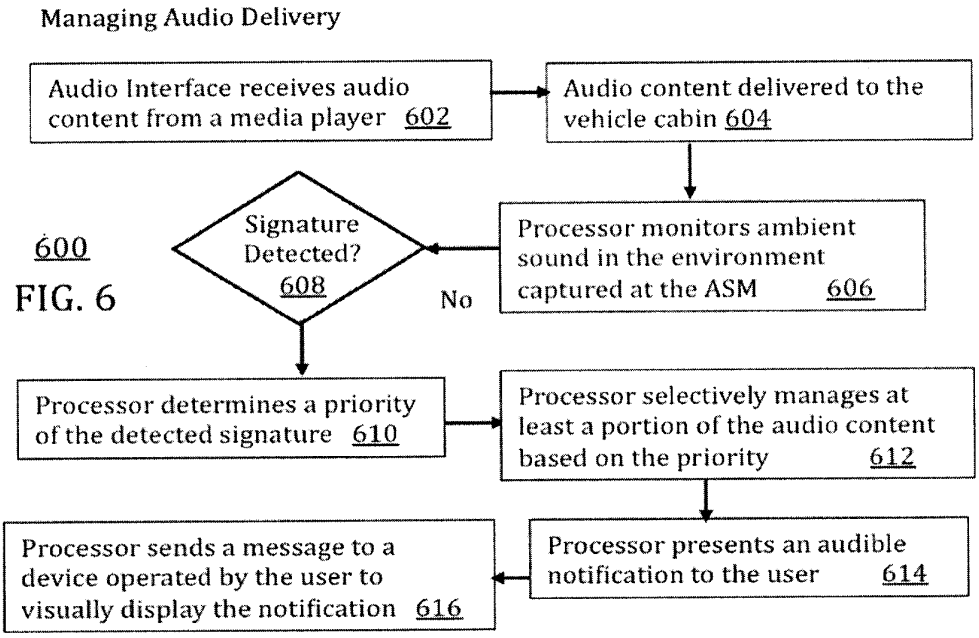

Managing Audio Delivery
Audio Interface receives audio content from a media player 602
Audio content delivered to the vehicle cabin 604
Processor monitors ambient sound in the environment captured at the ASM 606
Signature Detected? 608
No
Processor determines a priority of the detected signature 610
Processor selectively manages at least a portion of the audio content based on the priority 612
Processor presents an audible notification to the user 614
Processor sends a message to a device operated by the user to visually display the notification 616
600

FIG. 6

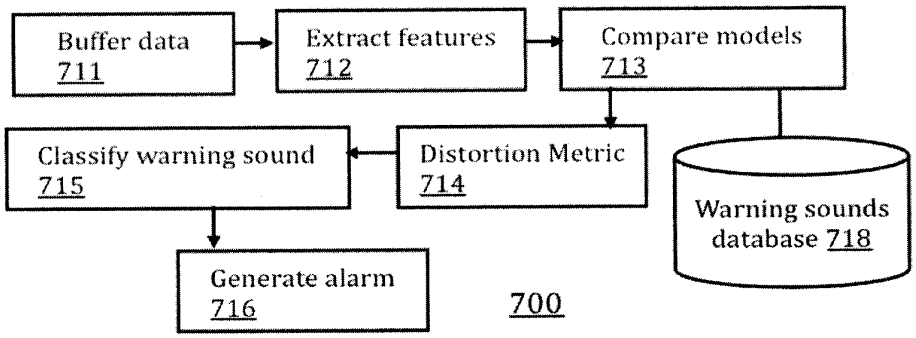
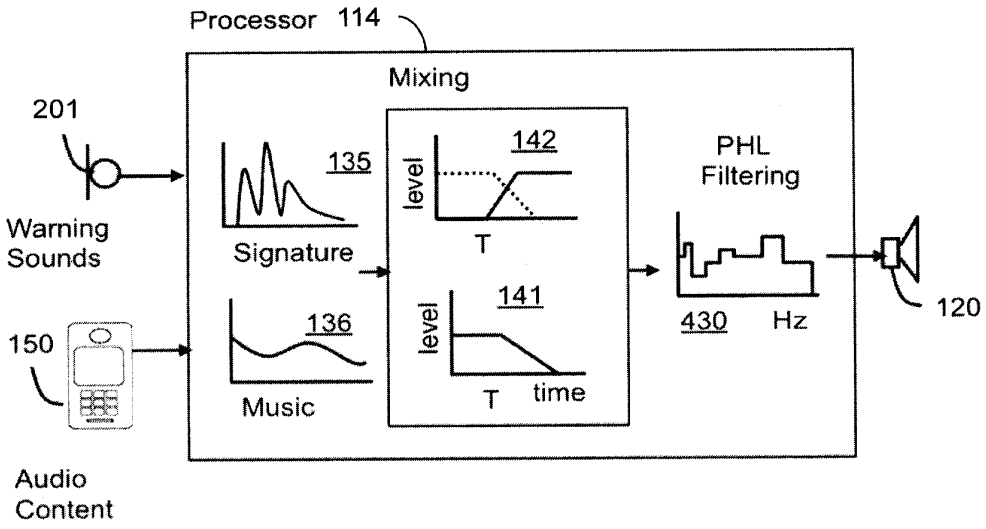

Buffer data 711
Extract features 712
Compare models 713
Distortion Metric 714
Classify warning sound 715
Generate alarm 716
Warning sounds database 718
700

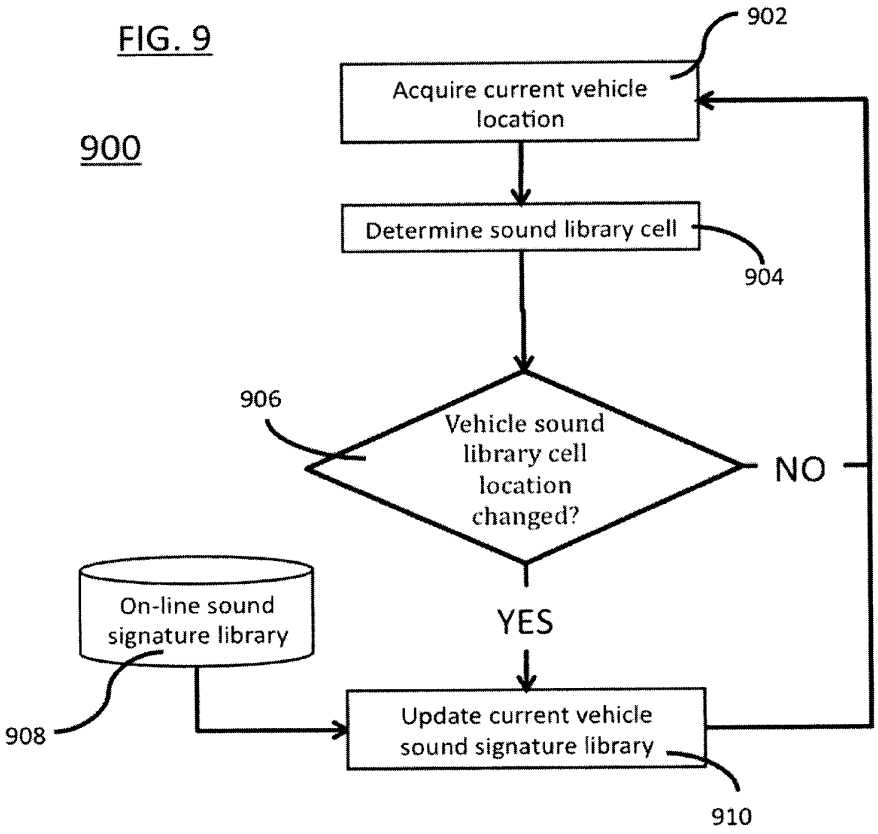
FIG. 7


Processor 114
Mixing
201
Warning Sounds
135
Signature
142
level
T
PHL Filtering
150
136
Music
141
level
T
time
430 Hz
120
Audio Content

800

FIG. 8

FIG. 9


900
902
Acquire current vehicle location
Determine sound library cell
904
906
Vehicle sound library cell location changed?
NO
YES
On-line sound signature library
908
Update current vehicle sound signature library
910

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US 12/21077

A. CLASSIFICATION OF SUBJECT MATTER
IPC(8) - H04B 1/00 (2012.01)
USPC - 381/86
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)
USPC 381/86

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
USPC 381/86, 87, 91, 94.1; 180/167

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
PubWEST (PGPB, USPT, EPAB, JPAB); Google.com; Google Scholar.
Search terms: vehicle, automobile, external microphone, ambient, exterior, noise, sound, warning, alarm, horn, siren, voice, volume, music, sound signature, verbal, spoken.

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 2007/0127734 A1 (Brulle-Drews) 07 June 2007 (07.06.2007) Entire document, especially Title, para. [0006], [0037], Fig. 4, 5.	1-27
Y	US 2008/0240458 A1 (Goldstein et al.) 2 October 2008 (02.10.2008) Entire document, especially para. [0002], [0009], [0011], [0026], [0028], [0036], [0040], [0045]-[004], [0052], [0059], [0061], [0065], Fig. 2, 8.	1-27

☐ Further documents are listed in the continuation of Box C. ☐

* 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 March 2012 (19.03.2012)	26 MAR 2012
Name and mailing address of the ISA/US Mail Stop PCT, Attn: ISA/US, Commissioner for Patents P.O. Box 1450, Alexandria, Virginia 22313-1450 Facsimile No. 571-273-3201	Authorized officer: Lee W. Young PCT Helpdesk: 571-272-4300 PCT OSP: 571-272-7774

Electronic Acknowledgement Receipt

EFS ID:	46072109
Application Number:	17215804
International Application Number:	
Confirmation Number:	9372
Title of Invention:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT2-16247186
Receipt Date:	29-JUN-2022
Filing Date:	29-MAR-2021
Time Stamp:	10:02:25
Application Type:	Utility under 35 USC 111(a)

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	Transmittal Letter	IDS-transmittal-ltr.pdf	121518 00768a2f178ebdacffb0feea3ad8ed9a9e623280	no	1

Warnings:

Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARDWARE-IDS-UPDATED.pdf	1036579 406a65f5c36d7527089fee7dd597d0ba89f5d8f1	no	11
Warnings:					
Information:					
3	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS-UPDATED.pdf	1037026 44b4870821cbb53acd7b962ab15aec72473cce4c	no	12
Warnings:					
Information:					
4	Information Disclosure Statement (IDS) Form (SB08)	MASTER-FOREIGN-NONPATENT-IDS.pdf	1035475 1d58e7c3bf3f49621567a1e08da3ab49a05d2e03	no	5
Warnings:					
Information:					
5	Foreign Reference	EP1385324A1-PDF.pdf	291998 fd44f66dc1543bbf582c8d89c4cae777488fcfff	no	15
Warnings:					
Information:					
6	Foreign Reference	EP1401240A1-PDF.pdf	269098 415e812ff51bde7a5bbcfea0ddc74575071e1f22	no	14
Warnings:					
Information:					
7	Foreign Reference	EP1519625B1-PDF.pdf	1060783 52f1291b06313730594dd459696b6ec0158d4382	no	27
Warnings:					
Information:					
8	Foreign Reference	EP1640972A1-PDF.pdf	219543 c59a9f5c0541d826a9c97e8648599bcc869ba635	no	15
Warnings:					
Information:					

9	Foreign Reference	JPH0877468-PDF.PDF	753639 8c7af5c8bc87c531d5920333f1968dd6c1519160	no	11
Warnings:					
Information:					
10	Foreign Reference	JPH10162283-PDF.PDF	1008899 21f5461a653457cc0cc84966600a4cb588cb28f6	no	15
Warnings:					
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11	Foreign Reference	WO2004114722A1-PDF.pdf	725963 bf5581f79d06ef7598680ed66e6f82a713bc0100	no	19
Warnings:					
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12	Foreign Reference	WO2006037156A1-PDF.pdf	1531590 348f0530551347acbea6bc41f27261c87be88ec1	no	37
Warnings:					
Information:					
13	Foreign Reference	WO2006054698-PDF.PDF	5448882 28699a5c6fd19433ee80f3e5f9470b2aea48f7cb	no	148
Warnings:					
Information:					
14	Foreign Reference	WO2007092660A1-PDF.pdf	159414 f1e779a0a3e51b76d6a7b22a5bb118212168a792	no	6
Warnings:					
Information:					
15	Foreign Reference	WO2008050583A1-PDF.pdf	2971807 88026f6ed1c5e16337df4097ac505bcea35770b2	no	67
Warnings:					
Information:					

16	Foreign Reference	WO2009023784-PDF.PDF	2445774 fcbe09b3b2fe64e4cae6738bc064fc7f8be833a6	no	37
Warnings:					
Information:					
17	Foreign Reference	WO2012097150A1-PDF.pdf	2797231 f7d2f88516c4c50354ef4dbced8bf4b4ab52aaf5	no	32
Warnings:					
Information:					
		Total Files Size (in bytes):	22915219		

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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

APPLICATION NO.	:	17/215,804 CONFIRMATION NO. 9372
APPLICATION OF	:	STEVEN WAYNE GOLDSTEIN
DATE FILED	:	29 MAR. 2021
EXAMINER	:	LAO, LUNSEE
TC/A.U.	:	2651
DOCKET	:	CONT2-16247186
TITLE	:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: June 29, 2022

Sir:

The attached extensive IDSs contain references filed in recent IPRs, the IPRs themselves and similar references filed in invalidity contentions filed in Court for similar inventive scope. Applicant has compiled the extensive list for the Examiner's consideration and apologizes ahead of time if any of the references are copies of previously considered related art.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,
TECHIYA LLC

Date: June 29, 2022

/John P. Keady/
John P. Keady, Ph.D (Physics), Ph.D (Comp. Sci)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
571-606-1291

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	LAO, LUNSEE
Attorney Docket Number	CONT2-16247186

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
	1	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
	5	7562020		2009-06-14	Le et al.	
	6	9135797		2015-09-15	Couper et al.	
	7	9123343		2015-09-01	Kurki-Suonio	
	8	7756285		2010-07-13	Sjursen et al.	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

	9	5903868		1999-05-11	Yuen et al.	
	10	3876843		1975-04-08	Moen	
	11	5208867		1993-05-04	Stites, III	
	12	6226389		2001-05-01	Lemelson et al.	
	13	8493204		2013-07-23	Wong et al.	
	14	6661901		2003-12-09	Svean et al.	
	15	7430299		2008-09-30	Armstrong et al.	
	16	7039195		2006-05-02	Svean	
	17	7107109		2006-09-12	Nathan et al.	
	18	4947440		1990-08-07	Bateman et al.	
	19	9037458		2015-05-19	Park et al.	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

	20	4054749		1977-10-18	Suzuki et al.	
	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
	25	6021325		2000-02-01	Hall	
	26	8750295		2014-06-10	Liron	
	27	7920557		2011-04-05	Moote	
	28	7158933		2007-01-02	Balan; Radu Victor	
	29	8611560		2013-12-17	Goldstein; Steven Wayne	
	30	5946050		1999-08-31	Wolff	

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	17215804
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	First Named Inventor	Steven Wayne Goldstein
	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

	31	7177433		2007-02-13	Sibbald	
	32	9053697		2015-06-09	Park	
	33	7903826		2011-03-08	Boersma	
	34	8150044		2012-04-03	Goldstein	
	35	8577062		2013-11-05	Goldstein	
	36	8774433		2014-07-08	Goldstein	
	37	8855343		2014-10-07	Usher	
	38	9191740		2015-11-17	McIntosh	
	39	9628896		2017-04-18	Ichimura	
	40	5577511		1996-11-26	Killion	
	41	6056698		2000-05-02	Iseberg	

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	Attorney Docket Number	CONT2-16247186

	42	5317273		1994-05-31	Hanson	
	43	8162846		2012-04-24	Epley	
	44	7280849		2007-10-09	Bailey	
	45	8045840		2011-10-25	Murata, et. al	
	46	6028514		2000-02-22	Lemelson	
	47	6408272		2002-06-18	White	

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	1	20030165246		2003-09-04	Kvaloy et al.	
	2	20050288057		2005-12-29	Lai et al.	
	3	20060195322		2006-08-31	Broussard et al.	

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	4	20070043563		2007-02-22	Comerford et al.	
	5	20010046304		2001-11-29	Rast	
	6	20070189544		2007-08-16	Rosenberg	
	7	20060092043		2006-05-04	Lagassey	
	8	20020106091		2002-08-08	Furst et al.	
	9	20040109668		2004-06-10	Stuckman	
	10	20020118798		2002-08-29	Langhart et al.	
	11	20040042103		2004-03-04	Mayer	
	12	20030161097		2003-08-28	Le et al.	
	13	20160104452		2016-04-14	Guan et al.	
	14	20040203351		2004-10-14	Shearer et al.	

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	Attorney Docket Number	CONT2-16247186

	15	20080165988		2008-07-10	Terlizzi et al.	
	16	20070291953		2007-12-20	Ngia et al.	
	17	20110096939		2011-04-28	Ichimura	
	18	20060067551		2006-03-30	Cartwright et al.	
	19	20090024234		2009-01-22	Archibald	
	20	20090010456		2009-01-08	Goldstein et al.	
	21	20070086600		2007-04-19	Boesen	
	22	20040190737		2004-09-30	Kuhnel et al.	
	23	20040196992		2004-10-07	Ryan	
	24	20050123146		2005-06-09	Voix et al.	
	25	20040125965		2004-07-01	Alberth Jr., et al.	

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	26	20060083395		2006-04-20	Allen et al.	
	27	20110264447		2011-10-27	Visser et al.	
	28	20100296668		2010-11-25	Lee et al.	
	29	20110293103		2011-12-01	Park et al.	
	30	20100061564		2010-03-11	Clemow et al.	
	31	20050058313		2005-03-17	Victorian, Thomas A.	
	32	20140122092		2019-03-19	Goldstein; Steven Wayne	
	33	20090286515		2009-05-05	Othmer; Konstantin	
	34	20060173563		2006-08-03	Borovitski; Michael D.	
	35	20090076821		2009-03-19	Brenner; Vadim	
	36	20050281423		2005-12-22	Armstrong	

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	Attorney Docket Number	CONT2-16247186

	37	20030130016		2003-07-10	Matsuura	
	38	20050207605		2005-09-22	Dehe	
	39	20070143820		2007-06-21	Pawlowski	
	40	20110055256		2011-03-03	Phillips	
	41	20060182287		2006-08-17	Schulein	
	42	20030198359		2003-10-23	Killion	
	43	20070086600		2007-04-19	Boesen	
	44	20060287014		2006-12-21	Matsuura	
	45	20100119077		2010-05-13	Platz	

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	Attorney Docket Number	CONT2-16247186

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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.

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-06-29
Name/Print	John P. Keady	Registration Number	56389

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Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	LAO, LUNSEE
Attorney Docket Number	CONT2-16247186

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	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
	15	6567524		2003-05-20	Svean et al.	
	16	6728385		2004-04-27	Kvaloy et al.	
	17	8140325		2012-03-20	Kanevsky	
	18	8983081		2015-03-17	Bayley	
	19	5327506		1994-07-05	Stites	

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	20	5933510		1999-08-03	Bryant	
	21	8086093		2011-12-27	Stuckman	
	22	7039585		2006-05-02	Wilmot	
	23	6606598		2003-08-12	Holthouse	
	24	6005525		1999-12-21	Kivela	
	25	RE38351		2003-12-16	Iseberg	
	26	7903825		2011-03-08	Melanson	
	27	8014553		2011-09-06	Radivojevic	
	28	8160261		2012-04-17	Schulein	
	29	6639987		2003-10-28	McIntosh	
	30	5923624		1999-07-13	Groeger	

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	31	8160273		2012-04-17	Visser	
	32	7464029		2008-12-09	Visser	
	33	7983907		2011-07-19	Visser	
	34	8254591		2012-08-28	Goldstein	
	35	8718305		2014-05-06	Usher	
	36	8798278		2014-08-05	Isabelle	
	37	8917894		2014-12-23	Goldstein	
	38	8625818		2014-01-07	Stultz	
	39	9491542		2016-11-08	Usher	
	40	8218784		2012-07-10	Schulein	
	41	6118877		2000-09-12	Lindemann	

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	42	7050592		2006-05-23	Iseberg	
	43	8401200		2013-03-19	Tiscareno	
	44	9196247		2015-11-24	Harada	
	45	7574917		2009-08-18	Von Dach	
	46	5276740		1994-01-04	Inanaga et al.	

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	3	20060204014		2006-09-14	Isenberg et al.	
	4	20050078838		2005-04-14	Simon	

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	5	20070036377		2007-02-15	Stirnemann	
	6	20020165719		2002-11-07	Wang	
	7	20060140425		2006-06-29	Berg	
	8	20030152359		2003-08-14	Kim	
	9	20070003090		2007-01-04	Anderson	
	10	20030165319		2003-09-04	Barber	
	11	20070255435		2007-11-01	Cohen	
	12	20040264938		2004-12-30	Felder	
	13	20060264176		2006-11-23	Hong	
	14	20040109579		2004-07-10	Izuchi	
	15	20060167687		2006-07-27	Kates	

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	16	20060083390		2006-04-20	Kaderavek	
	17	20050068171		2005-03-31	Kelliher	
	18	20050028212		2005-02-03	Laronne	
	19	20080145032		2008-06-19	Lindroos	
	20	20020098878		2002-07-25	Mooney	
	21	20060188075		2006-08-24	Peterson	
	22	20140023203		2014-01-23	Rotschild	
	23	20060188105		2006-08-24	Baskerville	
	24	20020076057		2002-06-20	Voix	
	25	20110116643		2011-05-19	Tiscareno	
	26	20150215701		2015-07-30	Usher	

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	27	20140163976		2014-06-12	Park	
	28	20080137873		2008-06-12	Goldstein	
	29	20090122996		2009-05-14	Klein	
	30	20040086138		2004-05-06	Kuth	
	31	20070021958		2008-01-25	Visser et al.	
	32	20020111798		2002-08-15	Huang	

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-06-29
Name/Print	John P. Keady	Registration Number	56389

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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.

Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT

(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	2651
Examiner Name	LAO, LUNSEE
Attorney Docket Number	CONT2-16247186

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	1	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00282, December 21, 2021	
	2	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00242, December 23, 2021	
	3	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00243, December 23, 2021	
	4	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00234, December 21, 2021	
	5	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00253, January 18, 2022	
	6	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00324, January 13, 2022	
	7	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00281, January 18, 2022	
	8	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00302, January 13, 2022	
	9	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00369, February 18, 2022	
	10	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00388, February 18, 2022	
	11	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-00410, February 18, 2022	

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	Filing Date	2021-03-29
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	Art Unit	2651
	Examiner Name	LAO, LUNSEE
	Attorney Docket Number	CONT2-16247186

	12	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-01078, June 9, 2022	
	13	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-01099, June 9, 2022	
	14	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-01106, June 9, 2022	
	15	SAMSUNG ELECTRONICS CO., LTD., and SAMSUNG ELECTRONICS, AMERICA, INC., v. STATON TECHIYA, LLC, IPR2022-01098, June 9, 2022	

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CERTIFICATION STATEMENT

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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).

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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	/John P. Keady/	Date (YYYY-MM-DD)	2022-06-29
Name/Print	John P. Keady	Registration Number	56389

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Electronic Acknowledgement Receipt

EFS ID:	46072175
Application Number:	17215804
International Application Number:	
Confirmation Number:	9372
Title of Invention:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT2-16247186
Receipt Date:	29-JUN-2022
Filing Date:	29-MAR-2021
Time Stamp:	10:07:29
Application Type:	Utility under 35 USC 111(a)

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)	MASTER-IPR-IDS.pdf	1034886 f185aa1a8b933af4679acb800b51b88539272202	no	5

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Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT2-16247186

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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
	1	6359993		2002-03-19	Brimhall	
	2	7072482		2006-07-04	Van Doorn et al.	
	3	7778434		2010-08-17	Juneau et al.	
	4	5524056		1996-06-04	Killion et al.	
	5	6415034		2002-07-02	Hietanen	
	6	8014553		2011-09-06	Radivojevic et al.	
	7	4088849		1978-05-09	Usami et al.	
	8	7209569		2007-04-24	Boesen	

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	9	6754359		2004-06-22	Svean et al.	
	10	6021207		2000-02-01	Puthuff et al.	
	11	5267321		1993-11-30	Langberg	
	12	RE38351		2003-12-16	Iseberg et al.	
	13	7477756		2009-01-13	Wickstrom et al.	
	14	7450730		2008-11-11	Bertg et al.	
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	1	20030165246		2003-09-04	Kvaloy et al.	

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	2	20080037801		2008-02-14	Alves et al.		
	3	20060204014		2006-09-14	Isenberg et al.		
	4	20050078838		2005-04-14	Simon		

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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 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.

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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.

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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:	44706133
Application Number:	17215804
International Application Number:	
Confirmation Number:	9372
Title of Invention:	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	150139
Filer:	John Patrick Keady
Filer Authorized By:	
Attorney Docket Number:	CONT2-16247186
Receipt Date:	10-JAN-2022
Filing Date:	29-MAR-2021
Time Stamp:	14:11:24
Application Type:	Utility under 35 USC 111(a)

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	Transmittal Letter	17215804_transmittal_ltr.pdf	125750 2703603a9034fefe6e90523f1e1e5739ccbff178	no	1

Warnings:

Information:					
2	Information Disclosure Statement (IDS) Form (SB08)	MASTER_SOFTWARE-IDS.pdf	1036626 14929f2b3a3a87233f3afe617826cd7abf280310	no	9
Warnings:					
Information:					
3	Information Disclosure Statement (IDS) Form (SB08)	MASTER_HARWARE-IDS.pdf	1035160 67be17fedba89b738d9167cd741cf5c115aa3fbe	no	6
Warnings:					
Information:					
4	Non Patent Literature	ref2-pdf2.pdf	3762561 39646ce913fa97e4a1efb547fcce8a48fc8e6d08	no	10
Warnings:					
Information:					
5	Non Patent Literature	ref1.pdf	1115022 01c5b583e9d89f0e10dda2f41631f1a7c8c19dda	no	2
Warnings:					
Information:					
6	Non Patent Literature	ref3.pdf	2858776 2418fbe7a8c340eaa35ef3922c589767859789c3	no	3
Warnings:					
Information:					
			Total Files Size (in bytes):	9933895	

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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

APPLICATION NO.	:	17/215,804	CONFIRMATION NO.	9372
APPLICATION OF	:	Steven Wayne Goldstein		

TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: January 10, 2022

Sir:

Please find attached:

- **IDS:** The attached IDS, consisting of two parts, list references that were cited in several IPRs for related Patents with some of the same inventors in the Staton Techiya LLC Patent Portfolio. Applicant appreciates the Examiner for taking the time to consider the extensive list for the present application. Attached are also three non-patent references cited. Should the Examiner have any questions, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

/John P. Keady/

John P. Keady
Ph.D. (Physics), Ph.D.(Comp. Sci.)
Assistant General Counsel: Intellectual Property
and Chief Technical Officer
Staton Techiya LLC
Reg. No. 56,389
Tel: 571-606-1291

PTO/SB/08a (02-18)
Approved for use through 11/30/2020. OMB 0651-0031
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
(Not for submission under 37 CFR 1.99)

Application Number	17215804
Filing Date	2021-03-29
First Named Inventor	Steven Wayne Goldstein
Art Unit	OPAP
Examiner Name	UNKNOWN
Attorney Docket Number	CONT2-16247186

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
	1	6804638		2004-10-12	Fiedler; Mark	
	2	6163508		2000-12-19	Kim et al.	
	3	6400652		2002-06-04	Goldberg et al.	
	4	7433714		2008-10-07	Howard et al.	
	5	7562020		2009-06-14	Le et al.	
	6	9135797		2015-09-15	Couper et al.	
	7	9123343		2015-09-01	Kurki-Suonio	
	8	7756285		2010-07-13	Sjursen et al.	

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	First Named Inventor	Steven Wayne Goldstein
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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT2-16247186

	9	5903868		1999-05-11	Yuen et al.	
	10	3876843		1975-04-08	Moen	
	11	5208867		1993-05-04	Stites, III	
	12	6226389		2001-05-01	Lemelson et al.	
	13	8493204		2013-07-23	Wong et al.	
	14	6661901		2003-12-09	Svean et al.	
	15	7430299		2008-09-30	Armstrong et al.	
	16	6567524		2003-05-20	Svean et al.	
	17	7107109		2006-09-12	Nathan et al.	
	18	4947440		1990-08-07	Bateman et al.	
	19	9037458		2015-05-19	Park et al.	

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	20	4054749		1977-10-18	Suzuki et al.	
	21	6298323		2001-10-02	Kaemmerer	
	22	6804643		2004-10-12	Kiss	
	23	6163338		2000-12-19	Johnson et al.	
	24	6748238		2004-06-08	Lau	
	25	6021325		2000-02-01	Hall	
	26	8750295		2014-06-10	Liron	
	27	7920557		2011-04-05	Moote	

If you wish to add additional U.S. Patent citation information please click the Add button. Add

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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
	1	20030165246		2003-09-04	Kvaloy et al.	

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	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT2-16247186

	2	20050288057		2005-12-29	Lai et al.	
	3	20060195322		2006-08-31	Broussard et al.	
	4	20070043563		2007-02-22	Comerford et al.	
	5	20010046304		2001-11-29	Rast	
	6	20070189544		2007-08-16	Rosenberg	
	7	20060092043		2006-05-04	Lagassey	
	8	20020106091		2002-08-08	Furst et al.	
	9	20040109668		2004-06-10	Stuckman	
	10	20020118798		2002-08-29	Langhart et al.	
	11	20040042103		2004-03-04	Mayer	
	12	20030161097		2003-08-28	Le et al.	

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	First Named Inventor	Steven Wayne Goldstein
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	Attorney Docket Number	CONT2-16247186

	13	20160104452		2016-04-14	Guan et al.	
	14	20040203351		2004-10-14	Shearer et al.	
	15	20080165988		2008-07-10	Terlizzi et al.	
	16	20070291953		2007-12-20	Ngia et al.	
	17	20110096939		2011-04-28	Ichimura	
	18	20060067551		2006-03-30	Cartwright et al.	
	19	20090024234		2009-01-22	Archibald	
	20	20090010456		2009-01-08	Goldstein et al.	
	21	20070086600		2007-04-19	Boesen	
	22	20040190737		2004-09-30	Kuhnel et al.	
	23	20040196992		2004-10-07	Ryan	

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	24	20050123146		2005-06-09	Voix et al.	
	25	20040125965		2004-07-01	Alberth Jr., et al.	
	26	20060083395		2006-04-20	Allen et al.	
	27	20110264447		2011-10-27	Visser et al.	
	28	20100296668		2010-11-25	Lee et al.	
	29	20110293103		2011-12-01	Park et al.	
	30	20100061564		2010-03-11	Clemow et al.	

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FOREIGN PATENT DOCUMENTS Remove

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]
	1	EP1519625	EP	A2		Victorian		
	2	2006037156	WO	A1		Mejia et al.		

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	Application Number	17215804
	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT2-16247186

NON-PATENT LITERATURE DOCUMENTS			Remove
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]
	1	Olwal, A. and Feiner S. Interaction Techniques Using Prosodic Features of Speech and Audio Localization. Proceedings of IUI 2005 (International Conference on Intelligent User Interfaces), San Diego, CA, Jan 9-12, 2005, p. 284-286.	
	2	Bernard Widrow, John R. Glover Jr., John M. McCool, John Kaunitz, Charles S. Williams, Robert H. Hearn, James R. Zeidler, Eugene Dong Jr, and Robert C. Goodlin, Adaptive Noise Cancelling: Principles and Applications, PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975	
	3	Mauro Dentino, John M. McCool, and Bernard Widrow, Adaptive Filtering in the Frequency Domain, PROCEEDINGS OF THE IEEE, VOL. 66, NO. 12, DECEMBER 1978	

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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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	Filing Date	2021-03-29
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	OPAP
	Examiner Name	UNKNOWN
	Attorney Docket Number	CONT2-16247186

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	/John P. Keady/	Date (YYYY-MM-DD)	2022-01-10
Name/Print	John P. Keady	Registration Number	56389

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APPLICATION NUMBER	FILING OR 371(C) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
17/215,804	03/29/2021	Steven Wayne Goldstein	CONT2-16247186

CONFIRMATION NO. 9372

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

PUBLICATION NOTICE

OC000000127002476

Title:METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

Publication No.US-2021-0219051-A1
Publication Date:07/15/2021

NOTICE OF PUBLICATION OF APPLICATION

The above-identified application will be electronically published as a patent application publication pursuant to 37 CFR 1.211, et seq. The patent application publication number and publication date are set forth above.

The publication may be accessed through the USPTO's publically available Searchable Databases via the Internet at www.uspto.gov. The direct link to access the publication is currently http://www.uspto.gov/patft/.

The publication process established by the Office does not provide for mailing a copy of the publication to applicant. A copy of the publication may be obtained from the Office upon payment of the appropriate fee set forth in 37 CFR 1.19(a)(1). Orders for copies of patent application publications are handled by the USPTO's Public Records Division. The Public Records Division can be reached by telephone at (571) 272-3150 or (800) 972-6382, by facsimile at (571) 273-3250, by mail addressed to the United States Patent and Trademark Office, Public Records Division, Alexandria, VA 22313-1450 or via the Internet.

In addition, information on the status of the application, including the mailing date of Office actions and the dates of receipt of correspondence filed in the Office, may also be accessed via the Internet through the Patent Electronic Business Center at www.uspto.gov using the public side of the Patent Application Information and Retrieval (PAIR) system. The direct link to access this status information is currently https://portal.uspto.gov/pair/PublicPair. Prior to publication, such status information is confidential and may only be obtained by applicant using the private side of PAIR.

Further assistance in electronically accessing the publication, or about PAIR, is available by calling the Patent Electronic Business Center at 1-866-217-9197.

Office of Data Managment, Application Assistance Unit (571) 272-4000, or (571) 272-4200, or 1-888-786-0101

PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 17/215,804

APPLICATION AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	80		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	350		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	400		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	14 minus 20 =	*	x 50 =	0.00	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	1 minus 3 =	*	x 240 =	0.00			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $310 ($155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	830		TOTAL	

APPLICATION AS AMENDED - PART II

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT A	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	RATE($)	ADDITIONAL FEE($)	OR	RATE($)	ADDITIONAL FEE($)
AMENDMENT B	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest found in the appropriate box in column 1.

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
17/215,804	03/29/2021	2656	830	CONT2-16247186	14	1

CONFIRMATION NO. 9372

150139
Akerman LLP
777 South Flagler Drive
Suite 1100 West Tower
West Palm Beach, FL 33401

FILING RECEIPT

OC000000124643979

Date Mailed: 04/07/2021

Receipt is acknowledged of this non-provisional utility patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF FIRST INVENTOR, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection.

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Inventor(s)

Steven Wayne Goldstein, Delray Beach, FL;
Marc Andre Boillot, Plantation, FL;
John Usher, Devon, UNITED KINGDOM;
Jason McIntosh, Sugar Hill, GA;

Applicant(s)

Staton Techiya LLC, Lighthouse Point, FL

Assignment For Published Patent Application

Staton Techiya LLC, Lighthouse Point, FL

Power of Attorney: None

Domestic Priority data as claimed by applicant

This application is a CIP of 16/247,186 01/14/2019
which is a CON of 13/956,767 08/01/2013 PAT 10182289
which is a CON of 12/170,171 07/09/2008 PAT 8526645
which is a CIP of 12/115,349 05/05/2008 PAT 8081780
which claims benefit of 60/916,271 05/04/2007

Foreign Applications for which priority is claimed (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.) - None.
Foreign application information must be provided in an Application Data Sheet in order to constitute a claim to foreign priority. See 37 CFR 1.55 and 1.76.

Permission to Access Application via Priority Document Exchange: Yes

Permission to Access Search Results: Yes

Applicant may provide or rescind an authorization for access using Form PTO/SB/39 or Form PTO/SB/69 as appropriate.

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The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 17/215,804**

Projected Publication Date: 07/15/2021

Non-Publication Request: No

Early Publication Request: No

** **SMALL ENTITY** **

Title

METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

Preliminary Class

381

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications: No

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Form Revision Date: March 1, 2019

Doc code: IDS
Doc description: Information Disclosure Statement (IDS) Filed

PTO/SB/08a (02-18)
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (**Not for submission under 37 CFR 1.99)**		
	Application Number	
	Filing Date	
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	
	Examiner Name	
	Attorney Docket Number	

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
	1	4809262		1989-02-28	Klose et al.	
	2	5259033		1993-11-02	Goodings et al.	
	3	5999828		1999-12-07	Sih et al.	
	4	6304648		2001-10-16	Chang	
	5	6381572		2002-04-30	Ishimitsu et al.	
	6	6647368		2003-11-11	Nemirovski	
	7	6754359		2004-06-22	Svean et al.	
	8	6870807		2005-03-22	Chan et al.	

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	9	7236580		2007-06-26	Sarkar et al.	
	10	7783054		2010-08-24	Ringlstetter et al.	
	11	7986802		2011-07-26	Ziller	
	12	8060366		2011-11-15	Maganti et al.	
	13	8275145		2012-09-25	Buck et al.	
	14	8380521		2013-02-19	Maganti et al.	
	15	8401178		2013-03-19	Chen et al.	
	16	8838184		2014-09-16	Burnett et al.	
	17	7039195		2006-05-02	Svean et al.	
	18	8081780		2011-12-20	Goldstein et al.	
	19	6728385		2004-04-27	Kvalo	

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	Attorney Docket Number	

	20	6738482		2004-05-18	Jaber	
	21	6081732		2000-06-27	Suvanen et al.	
	22	6169912		2001-01-02	Zuckerman	
	23	6570985		2003-05-27	Romesburg	
	24	7349353		2008-03-25	Guduru et al.	
	25	5131032		1992-07-14	Esaki et al.	
	26	6760453		2004-07-06	Banno	
	27	5963901		1999-10-05	Vanatalo et al.	
	28	6021207		2000-02-01	Puthuff	
	29	5692059		1997-11-25	Kruger	
	30	6118878		2000-09-12	Jones	

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	31	6631196		2003-10-07	Taenzer et al.	
	32	7817803		2010-10-19	Goldstein	
	33	8027481		2011-09-27	Beard	
	34	5796819		1998-08-18	Romesburg	
	35	7403608		2008-07-22	Auvray et al.	
	36	7003097		2006-02-21	Marchok et al.	
	37	6466666		2002-10-15	Eriksson	

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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
	1	20080037801		2008-02-14	Alves et al.	
	2	20010046304		2001-11-29	Rast	

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	First Named Inventor	Steven Wayne Goldstein
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	Examiner Name	
	Attorney Docket Number	

	3	20030112947		2003-06-19	Cohen	
	4	20040137969		2004-07-15	Nassimi	
	5	20050102133		2005-05-12	Rees	
	6	20060062395		2006-03-23	Klayman et al.	
	7	20070291953		2007-12-20	Ngia et al.	
	8	20080019539		2008-01-24	Patel et al.	
	9	20090147966		2009-06-11	McIntosh et al.	
	10	20120184337		2012-07-19	Burnett et al.	
	11	20130051543		2013-02-28	McDysan et al.	
	12	20140370838		2014-12-18	Kim	
	13	20150288823		2015-10-08	Burnett et al.	

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	14	20090010444		2009-01-08	Goldstein et al.	
	15	20050058313		2005-03-17	Victorian et al.	
	16	20050069161		2005-03-31	Kaltenbach et al.	
	17	20070189544		2007-06-16	Rosenberg	
	18	20040202340		2004-10-14	Armstrong et al.	
	19	20090034748		2009-02-05	Sibbald	
	20	20070036342		2007-02-15	Boillot et al.	
	21	20060067512		2006-03-30	Boillot et al.	

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	Filing Date	
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	
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CERTIFICATION STATEMENT

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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.

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Signature	/John P. Keady/	Date (YYYY-MM-DD)	2021-03-29
Name/Print	John P. Keady	Registration Number	56389

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT
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Application Number	
Filing Date	
First Named Inventor	Steven Wayne Goldstein
Art Unit	
Examiner Name	
Attorney Docket Number	

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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
	1	8600085		2013-12-03	Shaohai Chen, et al.	
	2	9584896		2017-02-28	Lethinal Kennedy	
	3	9270244		2016-02-23	John Usher, et al.	
	4	9936297		2018-04-03	George Joseph Dennis	
	5	9508335		2016-11-29	Benjamin D. Benattar, et al.	
	6	10499139		2019-12-03	Alaganandan Ganeshkumar	
	7	8798283		2014-08-05	Daniel M. Gauger, et al.	
	8	9953626		2018-04-24	Daniel M. Gauger, Jr., et al.	

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	Examiner Name	
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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
	1	20170345406		2017-11-30	Archiveades Georgiou, et al.	
	2	20140241553		2014-08-28	Victor Tiscareno, et al.	
	3	20070177741		2007-08-02	Matthew Williamson	
	4	20080205664		2008-08-28	Ju Yun Kim, et al.	
	5	20110103606		2011-05-05	Michael W. Silber	
	6	20170345406		2017-11-30	Archiveades Georgiou, et al.	

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	1	20080111004	KR		2008-12-22	요한 부르스마		
	2	2013501969	JP		2013-01-17	アーチビーディス ジョージョウアントニー ゼノノス		

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	Application Number	
	Filing Date	
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	
	Examiner Name	
	Attorney Docket Number	

	3	TWM568011U	TW		2018-10-01	劉上杰胡世昌		
	4	WO2008077981	WO		2008-07-03	Farzin Tahmassebi		
	5	ES2273616	ES		2007-12-16	Farzin Tahmassebi		
	6	CN105554610	CN		2019-01-04	刘崧赵剑		
	7	EP2963647	EP		2019-07-31	James M. Kirsch, et al.		
	8	CN203761556	CN		2014-08-06	许季明		
	9	CN105637892	CN		2020-03-13	D·S·布里格斯T·E·泰勒		
	10	JP6389232	JP		2018-09-12	ジョン ディー．et al.		

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)		
	Application Number	
	Filing Date	
	First Named Inventor	Steven Wayne Goldstein
	Art Unit	
	Examiner Name	
	Attorney Docket Number	

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CERTIFICATION STATEMENT

Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):

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☐ 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	/John P. Keady/	Date (YYYY-MM-DD)	2021-03-29
Name/Print	John P. Keady	Registration Number	56389

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION NO.	:	UNKNOWN	CONFIRMATION NO. UNKNOWN
APPLICATION OF	:	STEVEN WAYNE GOLDSTEIN	
DATE FILED	:	HEREWITH	
EXAMINER	:	UNKNOWN	
TC/A.U.	:	UNKNOWN	
DOCKET	:	CONT2-16247186	
TITLE :		METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

CONTINUATION IN PART TRANSMITTAL LETTER

Commissioner for Patents
P. O. Box 1450
Alexandria, VA 22313-1450

Date: March 29, 2021

Sir:

Please find attached a Continuation in Part application claiming priority to allowed U.S. Pat. App. No. 16/247,186. Although the present application is filed as a Continuation in Part it contains material from priority applications No. 12/170,171, 12/115,349 and 60/916,271 not originally included in the figures and specification of 16/247,186 although all were incorporated by reference in their entirety.

Should the Examiner believe that anything further is necessary, the Examiner is respectfully requested to contact the undersigned representative at the telephone number listed below.

Respectfully submitted,

TECHIYA LLC

Date: March 29, 2021

/John P. Keady/
John P. Keady
CTO and Assistant General Counsel IP
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Reg. No. 56,389
571-606-1291

METHOD AND DEVICE FOR IN-EAR CANAL ECHO SUPPRESSION

CROSS REFERENCE TO RELATED APPLICATIONS

[0001] This application is a Continuation in Part of U.S. Patent Application Serial No. 16/247,186, filed 14 Jan. 2019, which is a Continuation of U.S. Patent Application No. 13/956,767, filed on 1 August 2018, now U.S. Patent No. 10,182,289, which is a Continuation of U.S. Patent Application No. 12/170,171, filed on 9 July 2008, now U.S. Patent No. 8,526,645, which is a Continuation in Part of Application No. 12/115,349 filed on May 5, 2008, now U.S. Patent No. 8,081,780 which claims the priority benefit of Provisional Application No. 60/916,271 filed on May 4, 2007, the entire disclosure of all of which are incorporated herein by reference.

FIELD OF THE INVENTION

[0002] The present invention pertains to sound reproduction, sound recording, audio communications and hearing protection using earphone devices designed to provide variable acoustical isolation from ambient sounds while being able to audition both environmental and desired audio stimuli. Particularly, the present invention describes a method and device for suppressing echo in an ear-canal when capturing a user's voice when using an ambient sound microphone and an ear canal microphone.

BACKGROUND OF THE INVENTION

[0003] People use headsets or earpieces primarily for voice communications and music listening enjoyment. A headset or earpiece generally includes a microphone and a speaker for allowing the user to speak and listen. An ambient sound microphone mounted on the earpiece can capture ambient sounds in the environment; sounds that can include the user's voice. An ear canal microphone mounted internally on the earpiece can capture voice resonant within the ear canal; sounds generated when the user is speaking.

[0004] An earpiece that provides sufficient occlusion can utilize both the ambient sound microphone and the ear canal microphone to enhance the user's voice. An ear canal receiver mounted internal to the ear canal can loopback sound captured at the ambient sound microphone or the ear canal microphone to allow the user to listen to captured sound. If the earpiece is however not properly sealed within the ear canal, the ambient sounds can leak through into the ear canal and create an echo feedback condition with the ear canal microphone and ear canal receiver. In such cases, the feedback loop can generate an annoying "howling" sound that degrades the quality of the voice communication and listening experience.

SUMMARY OF THE INVENTION

[0005] Embodiments in accordance with the present invention provide a method and device for background noise control, ambient sound mixing and other audio control methods associated with an earphone. Note that although this application is filed as a continuation in part of U.S. Pat. App. No. 16/247,186, the subject matter material can be found in U.S. Patent Application No. 12/170,171, filed on 9 July 2008, now U.S. Patent No. 8,526,645, Application No. 12/115,349 filed on May 5, 2008, now U.S. Patent No. 8,081,780, and Application No. 60/916,271 filed on May 4, 2007, all of which were incorporated by reference in U.S. Pat. App. No. 16/247,186 and are incorporated by reference in their entirety herein.

[0006] In a first embodiment, a method for in-ear canal echo suppression control can include the steps of capturing an ambient acoustic signal from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, capturing in an ear canal an internal sound from at least one Ear Canal Microphone (ECM) to produce an electronic internal signal, measuring a background noise signal from the electronic ambient signal and the electronic internal signal, and capturing in the ear canal an internal sound from an Ear Canal Microphone (ECM) to produce an electronic internal signal. The electronic internal signal includes an echo of a spoken voice generated by a wearer of the earpiece. The echo in the electronic internal signal can be suppressed to produce a modified electronic internal signal containing primarily the spoken voice. A voice activity level can be generated for the spoken voice based on characteristics of the modified electronic internal signal and a level of the background noise signal. The electronic ambient signal and the electronic internal signal

can then be mixed in a ratio dependent on the background noise signal to produce a mixed signal without echo that is delivered to the ear canal by way of the ECR.

[0007] An internal gain of the electronic internal signal can be increased as background noise levels increase, while an external gain of the electronic ambient signal can be decreased as the background noise levels increase. Similarly, the internal gain of the electronic internal signal can be increased as background noise levels decrease, while an external gain of the electronic ambient signal can be increased as the background noise levels decrease. The step of mixing can include filtering the electronic ambient signal and the electronic internal signal based on a characteristic of the background noise signal. The characteristic can be a level of the background noise level, a spectral profile, or an envelope fluctuation.

[0008] At low background noise levels and low voice activity levels, the electronic ambient signal can be amplified relative to the electronic internal signal in producing the mixed signal. At medium background noise levels and voice activity levels, low frequencies in the electronic ambient signal and high frequencies in the electronic internal signal can be attenuated. At high background noise levels and high voice activity levels, the electronic internal signal can be amplified relative to the electronic ambient signal in producing the mixed signal.

[0009] The method can include adapting a first set of filter coefficients of a Least Mean Squares (LMS) filter to model an inner ear-canal microphone transfer function (ECTF). The voice activity level of the modified electronic internal signal can be monitored, and an adaptation of the first set of filter coefficients for the modified electronic internal signal can be frozen if the voice activity level is above a predetermined threshold. The voice activity level can be determined by an energy level characteristic and a frequency response characteristic. A second set of filter coefficients for a replica of the LMS filter can be generated during the freezing and substituted back for the first set of filter coefficients when the voice activity level is below another predetermined threshold. The modified electronic internal signal can be transmitted to another voice communication device and looped back to the ear canal.

[0010] In a second embodiment, a method for in-ear canal echo suppression control can include capturing an ambient sound from at least one Ambient Sound Microphone (ASM) to produce an electronic ambient signal, delivering audio content to an ear canal by way of an Ear Canal Receiver (ECR) to produce an acoustic audio content, capturing in the ear canal

by way of an Ear Canal Receiver (ECR) the acoustic audio content to produce an electronic internal signal, generating a voice activity level of a spoken voice in the presence of the acoustic audio content, suppressing an echo of the spoken voice in the electronic internal signal to produce a modified electronic internal signal, and controlling a mixing of the electronic ambient signal and the electronic internal signal based on the voice activity level. At least one voice operation of the earpiece can be controlled based on the voice activity level. The modified electronic internal signal can be transmitted to another voice communication device and looped back to the ear canal.

[0011] The method can include measuring a background noise signal from the electronic ambient signal and the electronic internal signal, and mixing the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal that is delivered to the ear canal by way of the ECR. An acoustic attenuation level of the earpiece and an audio content level reproduced can be accounted for when adjusting the mixing based on a level of the audio content, the background noise level, and an acoustic attenuation level of the earpiece. The electronic ambient signal and the electronic internal signal can be filtered based on a characteristic of the background noise signal. The characteristic can be a level of the background noise level, a spectral profile, or an envelope fluctuation. The method can include applying a first gain (G1) to the electronic ambient signal, and applying a second gain (G2) to the electronic internal signal. The first gain and second gain can be a function of the background noise level and the voice activity level.

[0012] The method can include adapting a first set of filter coefficients of a Least Mean Squares (LMS) filter to model an inner ear-canal microphone transfer function (ECTF). The adaptation of the first set of filter coefficients can be frozen for the modified electronic internal signal if the voice activity level is above a predetermined threshold. A second set of filter coefficients for a replica of the LMS filter can be adapted during the freezing. The second set can be substituted back for the first set of filter coefficients when the voice activity level is below another predetermined threshold. The adaptation of the first set of filter coefficients can then be unfrozen.

[0013] In a third embodiment, an earpiece to provide in-ear canal echo suppression can include an Ambient Sound Microphone (ASM) configured to capture ambient sound and produce an electronic ambient signal, an Ear Canal Receiver (ECR) to deliver audio content to an ear canal to produce an acoustic audio content, an Ear Canal Microphone (ECM)

configured to capture internal sound including spoken voice in an ear canal and produce an electronic internal signal, and a processor operatively coupled to the ASM, the ECM and the ECR. The audio content can be a phone call, a voice message, a music signal, or the spoken voice. The processor can be configured to suppress an echo of the spoken voice in the electronic internal signal to produce a modified electronic internal signal, generate a voice activity level for the spoken voice based on characteristics of the modified electronic internal signal and a level of the background noise signal, and mix the electronic ambient signal with the electronic internal signal in a ratio dependent on the background noise signal to produce a mixed signal that is delivered to the ear canal by way of the ECR. The processor can play the mixed signal back to the ECR for loopback listening. A transceiver operatively coupled to the processor can transmit the mixed signal to a second communication device.

[0014] A Least Mean Squares (LMS) echo suppressor can model an inner ear-canal microphone transfer function (ECTF) between the ASM and the ECM. A voice activity detector operatively coupled to the echo suppressor can adapt a first set of filter coefficients of the echo suppressor to model an inner ear-canal microphone transfer function (ECTF), and freeze an adaptation of the first set of filter coefficients for the modified electronic internal signal if the voice activity level is above a predetermined threshold. The voice activity detector during the freezing can also adapt a second set of filter coefficients for the echo suppressor, and substitute the second set of filter coefficients for the first set of filter coefficients when the voice activity level is below another predetermined threshold. Upon completing the substitution, the processor can unfreeze the adaptation of the first set of filter coefficients

BRIEF DESCRIPTION OF THE DRAWINGS

[0015] FIG. 1 is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

[0016] FIG. 2 is a block diagram of the earpiece in accordance with an exemplary embodiment;

[0017] FIG. 3 is a block diagram for an acoustic management module in accordance with an exemplary embodiment;

[0018] FIG. 4 is a schematic for the acoustic management module of FIG. 3 illustrating a mixing of an external microphone signal with an internal microphone signal as a function of

a background noise level and voice activity level in accordance with an exemplary embodiment;

[0019] FIG. 5 is a more detailed schematic of the acoustic management module of FIG. 3 illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

[0020] FIG. 6 is a block diagram of a system for in-ear canal echo suppression in accordance with an exemplary embodiment;

[0021] FIG. 7 is a schematic of a control unit for controlling adaptation of a first set and second set of filter coefficients of an echo suppressor for in-ear canal echo suppression in accordance with an exemplary embodiment;

[0022] FIG. 8 is a block diagram of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment;

[0023] FIG. 9 is a block diagram of a method for calculating background noise levels in accordance with an exemplary embodiment;

[0024] FIG. 10 is a block diagram for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment;

[0025] FIG. 11 is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment; and

[0026] FIG. 12 is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL) in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

[0027] The following description of at least one exemplary embodiment is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

[0028] Processes, techniques, apparatus, and materials as known by one of ordinary skill in the relevant art may not be discussed in detail but are intended to be part of the enabling description where appropriate, for example the fabrication and use of transducers.

[0029] In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

[0030] Note that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed for following figures.

[0031] Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

[0032] Various embodiments herein provide a method and device for automatically mixing audio signals produced by a pair of microphone signals that monitor a first ambient sound field and a second ear canal sound field, to create a third new mixed signal. An Ambient Sound Microphone (ASM) and an Ear Canal Microphone (ECM) can be housed in an earpiece that forms a seal in the ear of a user. The third mixed signal can be auditioned by the user with an Ear Canal Receiver (ECR) mounted in the earpiece, which creates a sound pressure in the occluded ear canal of the user. A voice activity detector can determine when the user is speaking and control an echo suppressor to suppress associated feedback in the ECR.

[0033] When the user engages in a voice communication, the echo suppressor can suppress feedback of the spoken voice from the ECR. The echo suppressor can contain two sets of filter coefficients; a first set that adapts when voice is not present and becomes fixed when voice is present, and a second set that adapts when the first set is fixed. The voice activity detector can discriminate between audible content, such as music, that the user is listening to, and spoken voice generated by the user when engaged in voice communication. The third mixed signal contains primarily the spoken voice captured at the ASM and ECM without echo, and can be transmitted to a remote voice communications system, such as a mobile phone, personal media player, recording device, walkie-talkie radio, etc. Before the ASM and ECM signals are mixed, they can be echo suppressed and subjected to different filters and at optional additional gains. This permits a single earpiece to provide full-duplex voice communication with proper or improper acoustic sealing.

[0034] The characteristic responses of the ASM and ECM filter can differ based on characteristics of the background noise and the voice activity level. In some exemplary embodiments, the filter response can depend on the measured Background Noise Level (BNL). A gain of a filtered ASM and a filtered ECM signal can also depend on the BNL. The (BNL) can be calculated using either or both the conditioned ASM and/or ECM signal(s). The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and can be weighted using a frequency-weighting system, e.g. to give an A-weighted SPL level (i.e. the high and low frequencies are attenuated before the level of the microphone signals are calculated).

[0035] At least one exemplary embodiment of the invention is directed to an earpiece for voice operated control. Reference is made to FIG. 1 in which an earpiece device, generally indicated as earpiece 100, is constructed and operates in accordance with at least one exemplary embodiment of the invention. As illustrated, earpiece 100 depicts an electro-acoustical assembly 113 for an in-the-ear acoustic assembly, as it would typically be placed in the ear canal 131 of a user 135. The earpiece 100 can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, open-fit device, or any other suitable earpiece type. The earpiece 100 can be partially or fully occluded in the ear canal, and is suitable for use with users having healthy or abnormal auditory functioning.

[0036] Earpiece 100 includes an Ambient Sound Microphone (ASM) 111 to capture ambient sound, an Ear Canal Receiver (ECR) 125 to deliver audio to an ear canal 131, and an Ear Canal Microphone (ECM) 123 to assess a sound exposure level within the ear canal 131. The earpiece 100 can partially or fully occlude the ear canal 131 to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal 131, and to form an acoustic seal with the walls 129 of the ear canal at a location 127 between the entrance 117 to the ear canal and the tympanic membrane (or ear drum) 133. Such a seal is typically achieved by means of a soft and compliant housing of assembly 113. Such a seal creates a closed cavity 131 of approximately 5cc between the in-ear assembly 113 and the tympanic membrane 133. As a result of this seal, the ECR (speaker) 125 is able to generate a full range frequency response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum resulting from the sound field at the entrance to the ear canal 131. This seal is also a basis for a sound isolating performance of the electro-acoustic assembly.

[0037] Located adjacent to the ECR 125, is the ECM 123, which is acoustically coupled to the (closed or partially closed) ear canal cavity 131. One of its functions is that of measuring the sound pressure level in the ear canal cavity 131 as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of the earpiece 100. In one arrangement, the ASM 111 can be housed in the assembly 113 to monitor sound pressure at the entrance to the occluded or partially occluded ear canal. All transducers shown can receive or transmit audio signals to a processor 121 that undertakes audio signal processing and provides a transceiver for audio via the wired or wireless communication path 119.

[0038] The earpiece 100 can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece 100 in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

[0039] The earpiece 100 can measure ambient sounds in the environment received at the ASM 111. Ambient sounds correspond to sounds within the environment such as the sound of traffic noise, street noise, conversation babble, or any other acoustic sound. Ambient sounds can also correspond to industrial sounds present in an industrial setting, such as, factory noise, lifting vehicles, automobiles, and robots to name a few.

[0040] The earpiece 100 can generate an Ear Canal Transfer Function (ECTF) to model the ear canal 131 using ECR 125 and ECM 123, as well as an Outer Ear Canal Transfer function (OETF) using ASM 111. For instance, the ECR 125 can deliver an impulse within the ear canal and generate the ECTF via cross correlation of the impulse with the impulse response of the ear canal. The earpiece 100 can also determine a sealing profile with the user's ear to compensate for any leakage. It also includes a Sound Pressure Level Dosimeter to estimate sound exposure and recovery times. This permits the earpiece 100 to safely administer and monitor sound exposure to the ear.

[0041] Referring to FIG. 2, a block diagram 200 of the earpiece 100 in accordance with an exemplary embodiment is shown. As illustrated, the earpiece 100 can include the processor 121 operatively coupled to the ASM 111, ECR 125, and ECM 123 via one or more Analog to Digital Converters (ADC) 202 and Digital to Analog Converters (DAC) 203. The processor

121 can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated storage memory 208 such as Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device 100. The processor 121 can also include a clock to record a time stamp.

[0042] As illustrated, the earpiece 100 can include an acoustic management module 201 to mix sounds captured at the ASM 111 and ECM 123 to produce a mixed sound. The processor 121 can then provide the mixed signal to one or more subsystems, such as a voice recognition system, a voice dictation system, a voice recorder, or any other voice related processor or communication device. The acoustic management module 201 can be a hardware component implemented by discrete or analog electronic components or a software component. In one arrangement, the functionality of the acoustic management module 201 can be provided by way of software, such as program code, assembly language, or machine language.

[0043] The memory 208 can also store program instructions for execution on the processor 121 as well as captured audio processing data and filter coefficient data. The memory 208 can be off-chip and external to the processor 121 and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor 121. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor 121 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0044] The earpiece 100 can include an audio interface 212 operatively coupled to the processor 121 and acoustic management module 201 to receive audio content, for example from a media player, cell phone, or any other communication device, and deliver the audio content to the processor 121. The processor 121 responsive to detecting spoken voice from the acoustic management module 201 can adjust the audio content delivered to the ear canal. For instance, the processor 121 (or acoustic management module 201) can lower a volume of the audio content responsive to detecting a spoken voice. The processor 121 by way of the ECM 123 can also actively monitor the sound exposure level inside the ear canal

and adjust the audio to within a safe and subjectively optimized listening level range based on voice operating decisions made by the acoustic management module 201.

[0045] The earpiece 100 can further include a transceiver 204 that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver 204 can also provide support for dynamic downloading over-the-air to the earpiece 100. It should be noted also that next generation access technologies can also be applied to the present disclosure.

[0046] The location receiver 232 can utilize common technology such as a common GPS (Global Positioning System) receiver that can intercept satellite signals and therefrom determine a location fix of the earpiece 100.

[0047] The power supply 210 can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece 100 and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply 210 to improve sensory input via haptic vibration. As an example, the processor 121 can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

[0048] The earpiece 100 can further represent a single operational device or a family of devices configured in a master-slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece 100 can be reused in different form factors for the master and slave devices.

[0049] FIG. 3 is a block diagram of the acoustic management module 201 in accordance with an exemplary embodiment. Briefly, the Acoustic management module 201 facilitates monitoring, recording and transmission of user-generated voice (speech) to a voice communication system. User-generated sound is detected with the ASM 111 that monitors a sound field near the entrance to a user's ear, and with the ECM 123 that monitors a sound field in the user's occluded ear canal. A new mixed signal 323 is created by filtering and mixing the ASM and ECM microphone signals. The filtering and mixing process is automatically controlled depending on the background noise level of the ambient sound field to enhance intelligibility of the new mixed signal 323. For instance, when the background noise level is high, the acoustic management module 201 automatically increases the level

of the ECM 123 signal relative to the level of the ASM 111 to create the new signal mixed 323. When the background noise level is low, the acoustic management module 201 automatically decreases the level of the ECM 123 signal relative to the level of the ASM 111 to create the new signal mixed 323

[0050] As illustrated, the ASM 111 is configured to capture ambient sound and produce an electronic ambient signal 426, the ECR 125 is configured to pass, process, or play acoustic audio content 402 (e.g., audio content 321, mixed signal 323) to the ear canal, and the ECM 123 is configured to capture internal sound in the ear canal and produce an electronic internal signal 410. The acoustic management module 201 is configured to measure a background noise signal from the electronic ambient signal 426 or the electronic internal signal 410, and mix the electronic ambient signal 426 with the electronic internal signal 410 in a ratio dependent on the background noise signal to produce the mixed signal 323. The acoustic management module 201 filters the electronic ambient signal 426 and the electronic internal 410 signal based on a characteristic of the background noise signal using filter coefficients stored in memory or filter coefficients generated algorithmically.

[0051] In practice, the acoustic management module 201 mixes sounds captured at the ASM 111 and the ECM 123 to produce the mixed signal 323 based on characteristics of the background noise in the environment and a voice activity level. The characteristics can be a background noise level, a spectral profile, or an envelope fluctuation. The acoustic management module 201 manages echo feedback conditions affecting the voice activity level when the ASM 111, the ECM 123, and the ECR 125 are used together in a single earpiece for full-duplex communication, when the user is speaking to generate spoken voice (captured by the ASM 111 and ECM 123) and simultaneously listening to audio content (delivered by ECR 125).

[0052] In noisy ambient environments, the voice captured at the ASM 111 includes the background noise from the environment, whereas, the internal voice created in the ear canal 131 captured by the ECM 123 has less noise artifacts, since the noise is blocked due to the occlusion of the earpiece 100 in the ear. It should be noted that the background noise can enter the ear canal if the earpiece 100 is not completely sealed. In this case, when speaking, the user's voice can leak through and cause an echo feedback condition that the acoustic management module 201 mitigates.

[0053] FIG. 4 is a schematic of the acoustic management module 201 illustrating a mixing of the electronic ambient signal 426 with the electronic internal signal 410 as a function of a

background noise level (BNL) and a voice activity level (VAL) in accordance with an exemplary embodiment. As illustrated, the acoustic management module 201 includes an Automatic Gain Control (AGC) 302 to measure background noise characteristics. The acoustic management module 201 also includes a Voice Activity Detector (VAD) 306. The VAD 306 can analyze either or both the electronic ambient signal 426 and the electronic internal signal 410 to estimate the VAL. As an example, the VAL can be a numeric range such as 0 to 10 indicating a degree of voicing. For instance, a voiced signal can be predominately periodic due to the periodic vibrations of the vocal cords. A highly voiced signal (e.g., vowel) can be associated with a high level, and a non-voiced signal (e.g., fricative, plosive, consonant) can be associated with a lower level.

[0054] The acoustic management module 201 includes a first gain (G1) 304 applied to the AGC processed electronic ambient signal 426. A second gain (G2) 308 is applied to the VAD processed electronic internal signal 410. The acoustic management module 201 applies the first gain (G1) 304 and the second gain (G2) 308 as a function of the background noise level and the voice activity level to produce the mixed signal 323, where

$$G1 = f(BNL)+f(VAL) \text{ and } G2=f(BNL)+f(VAL)$$

[0055] As illustrated, the mixed signal 323 is the sum 310 of the G1 scaled electronic ambient signal and the G2 scaled electronic internal signal. The mixed signal 323 can then be transmitted to a second communication device (e.g. second cell phone, voice recorder, etc.) to receive the enhanced voice signal. The acoustic management module 201 can also play the mixed signal 323 back to the ECR for loopback listening. The loopback allows the user to hear himself or herself when speaking, as though the earpiece 100 and associated occlusion effect were absent. The loopback can also be mixed with the audio content 321 based on the background noise level, the VAL, and audio content level. The acoustic management module 201 can also account for an acoustic attenuation level of the earpiece, and account for the audio content level reproduced by the ECR when measuring background noise characteristics. Echo conditions created as a result of the loopback can be mitigated to ensure that the voice activity level is accurate.

[0056] FIG. 5 is a more detailed schematic of the acoustic management module 201 illustrating a mixing of an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary

embodiment. In particular, the gain blocks for G1 and G2 of FIG. 4 are a function of the BNL and the VAL and are shown in greater detail. As illustrated, the AGC produces a BNL that can be used to set a first gain 322 for the processed electronic ambient signal 311 and a second gain 324 for the processed electronic internal signal 312. For instance, when the BNL is low (<70 dBA), gain 322 is set higher relative to gain 324 so as to amplify the electronic ambient signal 311 in greater proportion than the electronic internal signal 312. When the BNL is high (>85 dBA), gain 322 is set lower relative to gain 324 so as to attenuate the electronic ambient signal 311 in greater proportion than the electronic internal signal 312. The mixing can be performed in accordance with the relation:

Mixed signal=(1-β)*electronic ambient signal+(β)*electronic internal signal

where (1-β) is an external gain, (β) is an internal gain, and the mixing is performed with $0<\beta<1$.

[0057] As illustrated, the VAD produces a VAL that can be used to set a third gain 326 for the processed electronic ambient signal 311 and a fourth gain 328 for the processed electronic internal signal 312. For instance, when the VAL is low (e.g., 0-3), gain 326 and gain 328 are set low so as to attenuate the electronic ambient signal 311 and the electronic internal signal 312 when spoken voice is not detected. When the VAL is high (e.g., 7-10), gain 326 and gain 328 are set high so as to amplify the electronic ambient signal 311 and the electronic internal signal 312 when spoken voice is detected.

[0058] The gain scaled processed electronic ambient signal 311 and the gain scaled processed electronic internal signal 312 are then summed at adder 320 to produce the mixed signal 323. The mixed signal 323, as indicated previously, can be transmitted to another communication device, or as loopback to allow the user to hear his or her self.

[0059] FIG. 6 is an exemplary schematic of an operational unit 600 of the acoustic management module for in-ear canal echo suppression in accordance with an embodiment. The operational unit 600 may contain more or less than the number of components shown in the schematic. The operational unit 600 can include an echo suppressor 610 and a voice decision logic 620.

[0060] The echo suppressor 610 can be a Least Mean Squares (LMS) or Normalized Least Mean Squares (NLMS) adaptive filter that models an ear canal transfer function (ECTF)

between the ECR 125 and the ECM 123. The echo suppressor 610 generates the modified electronic signal, $e(n)$, which is provided as an input to the voice decision logic 620; e(n) is also termed the error signal $e(n)$ of the echo suppressor 610. Briefly, the error signal $e(n)$ 412 is used to update the filter H(w) to model the ECTF of the echo path. The error signal $e(n)$ 412 closely approximates the user's spoken voice signal $u(n)$ 607 when the echo suppressor 610 accurately models the ECTF.

[0061] In the configuration shown the echo suppressor 610 minimizes the error between the filtered signal, $\tilde{y}(n)$, and the electronic internal signal, z(n), in an effort to obtain a transfer function H' which is a best approximation to the H(w) (i.e., ECTF). H(w) represents the transfer function of the ear canal and models the echo response. ($z(n)=u(n)+y(n)+v(n)$, where $u(n)$ is the spoken voice 607, $y(n)$ is the echo 609, and $v(n)$ is background noise (if present, for instance due to improper sealing).)

[0062] During operation, the echo suppressor 610 monitors the mixed signal 323 delivered to the ECR 125 and produces an echo estimate $\tilde{y}(n)$ of an echo $y(n)$ 609 based on the captured electronic internal signal 410 and the mixed signal 323. The echo suppressor 610, upon learning the ECTF by an adaptive process, can then suppress the echo $y(n)$ 609 of the acoustic audio content 603 (e.g., output mixed signal 323) in the electronic internal signal $z(n)$ 410. It subtracts the echo estimate $\tilde{y}(n)$ from the electronic internal signal 410 to produce the modified electronic internal signal $e(n)$ 412.

[0063] The voice decision logic 620 analyzes the modified electronic signal 412 e(n) and the electronic ambient signal 426 to produce a voice activity level 622, α. The voice activity level α identifies a probability that the user is speaking, for example, when the user is using the earpiece for two way voice communication. The voice activity level 622 can also indicate a degree of voicing (e.g., periodicity, amplitude), When the user is speaking, voice is captured externally (such as from acoustic ambient signal 424) by the ASM 111 in the ambient environment and also by the ECM 123 in the ear canal. The voice decision logic provides the voice activity level α to the acoustic management module 201 as an input parameter for mixing the ASM 111 and ECM 123 signals. Briefly referring back to FIG. 4, the acoustic management module 201 performs the mixing as a function of the voice activity level α and the background noise level (see $G=f(BNL)+f(VAL)$).

[0064] For instance, at low background noise levels and low voice activity levels, the acoustic management module 201 amplifies the electronic ambient signal 426 from the ASM 111 relative to the electronic internal signal 410 from the ECM 123 in producing the mixed

signal 323. At medium background noise levels and medium voice activity levels, the acoustic management module 201 attenuates low frequencies in the electronic ambient signal 426 and attenuates high frequencies in the electronic internal signal 410. At high background noise levels and high voice activity levels, the acoustic management module 201 amplifies the electronic internal signal 410 from the ECM 123 relative to the electronic ambient signal 426 from the ASM 111 in producing the mixed signal. The acoustic management module 201 can additionally apply frequency specific filters based on the characteristics of the background noise.

[0065] FIG. 7 is a schematic of a control unit 700 for controlling adaptation of a first set (736) and a second set (738) of filter coefficients of the echo suppressor 610 for in-ear canal echo suppression in accordance with an exemplary embodiment. Briefly, the control unit 700 illustrates a freezing (fixing) of weights upon detection of spoken voice. The echo suppressor resumes weight adaptation when e(n) is low, and freezes weights when e(n) is high signifying a presence of spoken voice.

[0066] When the user is not speaking, the ECR 125 can pass through ambient sound captured at the ASM 111, thereby allowing the user to hear environmental ambient sounds. As previously discussed, the echo suppressor 610 models an ECTF and suppresses an echo of the mixed signal 323 that is looped back to the ECR 125 by way of the ASM 111 (see dotted line Loop Back path). When the user is not speaking, the echo suppressor continually adapts to model the ECTF. When the ECTF is properly modeled, the echo suppressor 610 produces a modified internal electronic signal $e(n)$ that is low in amplitude level (i.e., low in error). The echo suppressor adapts the weights to keep the error signal low. When the user speaks, the echo suppressor however initially produces a high-level e(n) (e.g., the error signal increases). This happens since the speaker's voice is uncorrelated with the audio signal played out the ECR 125, which disrupts the echo suppressor's ECTF modeling ability.

[0067] The control unit 700 upon detecting a rise in e(n), freezes the weights of the echo suppressor 610 to produce a fixed filter H'(w) fixed 738. Upon detecting the rise in e(n) the control unit adjusts the gain 734 for the ASM signal and the gain 732 for the mixed signal 323 that is looped back to the ECR 125. The mixed signal 323 fed back to the ECR 125 permits the user to hear themselves speak. Although the weights are frozen when the user is speaking, a second filter H'(w) 736 continually adapts the weights for generating a second e(n) that is used to determine a presence of spoken voice. That is, the control unit 700

monitors the second error signal e(n) produced by the second filter 736 for monitoring a presence of the spoken voice.

[0068] The first error signal e(n) (in a parallel path) generated by the first filter 738 is used as the mixed signal 323. The first error signal contains primarily the spoken voice since the ECTF model has been fixed due to the weights. That is, the second (adaptive) filter is used to monitor a presence of spoken voice, and the first (fixed) filter is used to generate the mixed signal 323.

[0069] Upon detecting a fall of e(n), the control unit restores the gains 734 and 732 and unfreezes the weights of the echo suppressor, and the first filter H'(w) returns to being an adaptive filter. The second filter H'(w) 736 remains on stand-by until spoken voice is detected, and at which point, the first filter H'(w) 738 goes fixed, and the second filter H'(w) 736 begins adaptation for producing the e(n) signal that is monitored for voice activity. Notably, the control unit 700 monitors e(n) from the first filter 738 or the second filter 736 for changes in amplitude to determine when spoken voice is detected based on the state of voice activity.

[0070] FIG. 8 is a block diagram 800 of a method for an audio mixing system to mix an external microphone signal with an internal microphone signal based on a background noise level and voice activity level in accordance with an exemplary embodiment.

[0071] As illustrated the mixing circuitry 816 (shown in center) receives an estimate of the background noise level 812 for mixing either or both the right earpiece ASM signal 802 and the left earpiece ASM signal 804 with the left earpiece ECM signal 806. (The right earpiece ECM signal can be used similarly.) An operating mode selection system 814 selects a switching 808 (e.g., 2-in, 1-out) between the left earpiece ASM signal 804 and the right earpiece ASM signal 802. As indicated earlier, the ASM signals and ECM signals can be first amplified with a gain system and then filtered with a filter system (the filtering may be accomplished using either analog or digital electronics or both). The audio input signals 802, 804, and 806 are therefore taken after this gain and filtering process, if any gain and filtering are used.

[0072] The Acoustic Echo Cancellation (AEC) system 810 can be activated with the operating mode selection system 814 when the mixed signal audio output 828 is reproduced with the ECR 125 in the same ear as the ECM 123 signal used to create the mixed signal audio output 828. The acoustic echo cancellation platform 810 can also suppress an echo of a spoken voice generated by the wearer of the earpiece 100. This ensures against acoustic

feedback ("howlback").

[0073] The Voice Activated System (VOX) 818 in conjunction with a de-bouncing circuit 822 activates the electronic switch 826 to control the mixed signal output 828 from the mixing circuitry 816; the mixed signal is a combination of the left ASM signal 804 or right ASM signal 802, with the left ECM 806 signal. Though not shown, the same arrangement applies for the other earphone device for the right ear, if present. Note that earphones can be used in both ears simultaneously. In a contra-lateral operating mode, as selected by operating mode selection system 814, the ASM and ECM signal are taken from opposite earphone devices, and the mix of these signals is reproduced with the ECR in the earphone that is contra-lateral to the ECM signal, and the same as the ASM signal.

[0074] For instance, in the contra-lateral operating mode, the ASM signal from the Right earphone device is mixed with the ECM signal from the left earphone device, and the audio signal corresponding to a mix of these two signals is reproduced with the Ear Canal Receiver (ECR) in the Right earphone device. The mixed signal audio output 828 therefore can contain a mix of the ASM and ECM signals when the user's voice is detected by the VOX. This mixed signal audio output can be used in loopback as a user Self-Monitor System to allow the user to hear their own voice as reproduced with the ECR 125, or it may be transmitted to another voice system, such as a mobile phone, walkie-talkie radio etc. The VOX system 818 that activates the switch 826 may be one a number of VOX embodiments.

[0075] In a particular operating mode, specified by unit 814, the conditioned ASM signal is mixed with the conditioned ECM signal with a ratio dependent on the BNL using audio signal mixing circuitry and the method described in either FIG. 10 or FIG. 11. As the BNL increases, then the ASM signal is mixed with the ECM signal with a decreasing level. When the BNL is above a particular value, then a minimal level of the ASM signal is mixed with the ECM signal. When the VOX switch 618 is active, the mixed ASM and ECM signals are then sent to mixed signal output 828. The switch de-bouncing circuit 826 ensures against the VOX 818 rapidly closing on and off (sometimes called chatter). This can be achieved with a timing circuit using digital or analog electronics. For instance, with a digital system, once the VOX has been activated, a time starts to ensure that the switch 826 is not closed again within a given time period, e.g. 100 ms. The delay unit 824 can improve the sound quality of the mixed signal audio output 828 by compensating for any latency in voice detection by the VOX system 818. In some exemplary embodiments, the switch debouncing circuit 822 can be dependent by the BNL. For instance, when the BNL is high (e.g. above 85 dBA), the

de-bouncing circuit can close the switch 826 sooner after the VOX output 818 determines that no user speech (e.g. spoken voice) is present.

[0076] FIG. 9 is a block diagram of a method 920 for calculating background noise levels in accordance with an exemplary embodiment. Briefly, the background noise levels can be calculated according to different contexts, for instance, if the user is talking while audio content is playing, if the user is talking while audio content is not playing, if the user is not talking but audio content is playing, and if the user is not talking and no audio content is playing. For instance, the system takes as its inputs either the ECM and/or ASM signal, depending on the particular system configuration. If the ECM signal is used, then the measured BNL accounts for an acoustic attenuation of the earpiece and a level of reproduced audio content.

[0077] As illustrated, modules 922-928 provide exemplary steps for calculating a base reference background noise level. The ECM or ASM audio input signal 922 can be buffered 923 in real-time to estimate signal parameters. An envelope detector 924 can estimate a temporal envelope of the ASM or ECM signal. A smoothing filter 925 can minimize abruptions in the temporal envelope. (A smoothing window 926 can be stored in memory). An optional peak detector 927 can remove outlier peaks to further smooth the envelope. An averaging system 928 can then estimate the average background noise level (BNL_1) from the smoothed envelope.

[0078] If at step 929, it is determined that the signal from the ECM was used to calculate the BNL_1, an audio content level 932 (ACL) and noise reduction rating 933 (NRR) can be subtracted from the BNL_1 estimate to produce the updated BNL 931. This is done to account for the audio content level reproduced by the ECR 125 that delivers acoustic audio content to the earpiece 100, and to account for an acoustic attenuation level (i.e. Noise Reduction Rating 933) of the earpiece. For example, if the user is listening to music, the acoustic management module 201 takes into account the audio content level delivered to the user when measuring the BNL. If the ECM is not used to calculate the BNL at step 929, the previous real-time frame estimate of the BNL 930 is used.

[0079] At step 936, the acoustic management module 201 updates the BNL based on the current measured BNL and previous BNL measurements 935. For instance, the updated BNL 937 can be a weighted estimate 934 of previous BNL estimates according to BNL=2*previous BNL+(1-w)*current BNL, where 0<W<1. The BNL can be a slow time weighted average of the level of the ASM and/or ECM signals, and may be weighted using a

frequency-weighting system, e.g. to give an A-weighted SPL level.

[0080] FIG. 10 is a block diagram 1040 for mixing an external microphone signal with an internal microphone signal based on a background noise level to produce a mixed output signal in accordance with an exemplary embodiment. The block diagram can be implemented by the acoustic management module 201 or the processor 121. In particular, FIG. 10 primarily illustrates the selection of microphone filters based on the background noise level. The microphone filters are used to condition the external and internal microphone signals before mixing.

[0081] As shown, the filter selection module 1045 can select one or more filters to apply to the microphone signals before mixing. For instance, the filter selection module 1045 can apply an ASM filter 1048 to the ASM signal 1047 and an ECM filter 1051 to the ECM signal 1052 based on the background noise level 1042. The ASM and ECM filters can be retrieved from memory based on the characteristics of the background noise. An operating mode 1046 can determine whether the ASM and ECM filters are look-up curves 1043 from memory or filters whose coefficients are determined in real-time based on the background noise levels.

[0082] Prior to mixing with summing unit 1049 to produce output signal 1050, the ASM signal 1047 is filtered with ASM filter 1048, and the ECM signal 1052 is filtered with ECM filter 1051. The filtering can be accomplished by a time-domain transversal filter (FIR-type filter), an IIR-type filter, or with frequency-domain multiplication. The filter can be adaptive (i.e. time variant), and the filter coefficients can be updated on a frame-by-frame basis depending on the BNL. The filter coefficients for a particular BNL can be loaded from computer memory using pre-defined filter curves 1043, or can be calculated using a predefined algorithm 1044, or using a combination of both (e.g. using an interpolation algorithm to create a filter curve for both the ASM filter 1048 and ECM filter 1051 from predefined filters).

[0083] FIG. 11 is a block diagram for an analog circuit for mixing an external microphone signal with an internal microphone signal based on a background noise level in accordance with an exemplary embodiment.

[0084] In particular, FIG. 11 shows a method 1160 for the filtering of the ECM and ASM signals using analog electronic circuitry prior to mixing. The analog circuit can process both the ECM and ASM signals in parallel; that is, the analog components apply to both the ECM and ASM signals. In one exemplary embodiment, the input audio signal 1161 (e.g., ECM

signal, ASM signal) is first filtered with a fixed filter 1162. The filter response of the fixed filter 1162 approximates a low-pass shelf filter when the input signal 1161 is an ECM signal, and approximates a high-pass filter when the input signal 1161 is an ASM signal. In an alternate exemplary embodiment, the filter 1162 is a unity-pass filter (i.e. no spectral attenuation) and the gain units G1, G2 etc instead represent different analog filters. As illustrated, the gains are fixed, though they may be adapted in other embodiments. Depending on the BNL 1169, the filtered signal is then subjected to one of three gains; G1 1163, G2 1164, or G3 1165. (The analog circuit can include more or less than the number of gains shown.)

[0085] For low BNLs (e.g. when BNL<L 1170, where L1 is a predetermined level threshold 1171), a G1 is determined for both the ECM signal and the ASM signal. The gain G1 for the ECM signal is approximately zero; i.e. no ECM signal would be present in the output signal 1175. For the ASM input signal, G1 would be approximately unity for low BNL.

[0086] For medium BNLs (e.g. when BNL<L2 1172, where L2 is a predetermined level threshold 1173), a G2 is determined for both the ECM signal and the ASM signal. The gain G2 for the ECM signal and the ASM signal is approximately the same. In another embodiment, the gain G2 can be frequency dependent so as to emphasize low frequency content in the ECM and emphasize high frequency content in the ASM signal in the mix. For high BNL; G3 1165 is high for the ECM signal, and low for the ASM signal. The switches 1166, 1167, and 1168 ensure that only one gain channel is applied to the ECM signal and ASM signal. The gain scaled ASM signal and ECM signal are then summed at junction 1174 to produce the mixed output signal 1175.

[0087] Examples of filter response curves for three different BNL are shown in FIG. 12, which is a table illustrating exemplary filters suitable for use with an Ambient Sound Microphone (ASM) and Ear Canal Microphone (ECM) based on measured background noise levels (BNL).

[0088] The basic trend for the ASM and ECM filter response at different BNLs is that at low BNLs (e.g. <60 dBA), the ASM signal is primarily used for voice communication. At medium BNL; ASM and ECM are mixed in a ratio depending on the BNL, though the ASM filter can attenuate low frequencies of the ASM signal, and attenuate high frequencies of the ECM signal. At high BNL (e.g. >85 dB), the ASM filter attenuates most al the low frequencies of the ASM signal, and the ECM filter attenuates most all the high frequencies of the ECM signal. In another embodiment of the Acoustic Management System, the ASM and ECM

filters may be adjusted by the spectral profile of the background noise measurement. For instance, if there is a large Low Frequency noise in the ambient sound field of the user, then the ASM filter can reduce the low-frequencies of the ASM signal accordingly, and boost the low-frequencies of the ECM signal using the ECM filter.

[0089] Where applicable, the present embodiments of the invention can be realized in hardware, software or a combination of hardware and software. Any kind of computer system or other apparatus adapted for carrying out the methods described herein are suitable. A typical combination of hardware and software can be a mobile communications device with a computer program that, when being loaded and executed, can control the mobile communications device such that it carries out the methods described herein. Portions of the present method and system may also be embedded in a computer program product, which comprises all the features enabling the implementation of the methods described herein and which when loaded in a computer system, is able to carry out these methods.

[0090] While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

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First Named Inventor/Applicant Name:	Steven Wayne Goldstein
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Basic Filing:				
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UTILITY SEARCH FEE	2111	1	350	350
UTILITY EXAMINATION FEE	2311	1	400	400
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
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Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
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1	Application Data Sheet	WebADS.pdf	162146 07e021a8a3fb39c8f9dde4d247beb5975f8db69d	no	9
Warnings:					
Information:					
2	Oath or Declaration filed	16247186-OATHS.pdf	360352 dc8f8c4becabfcdd0873d762a4b2a34de9073ea8	no	4
Warnings:					
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3	Abstract	CONT2-16247186-ABSTRACT.pdf	89437 80795af87cb53dc7398321e003023ab17335570b	no	1
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4	Claims	CONT2-16247186-CLAIMS-DUALNC.pdf	96366 0b40f62900ac41659391cca059af8d890e6116a8	no	2
Warnings:					
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5	Drawings-other than black and white line drawings	CONT2-16247186-figs.pdf	1257312 9d810c2d6e5526545eb1ca8ddc4a03edd24d2dd3	no	10
Warnings:					
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6	Information Disclosure Statement (IDS) Form (SB08)	CONT2-16247186-IDS.pdf	561583 4b825679fdaafbf2e8f98468ea3b9c451259d992	no	9
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7	Information Disclosure Statement (IDS) Form (SB08)	CONT-14017711_IDS2.pdf	1040678 646ac33d54d3f0b6e47668cc1615c98741aeb5de	no	6
Warnings:					
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8	Transmittal Letter	CONT2-16247186-ltr.pdf	100020 fbe9960619c3c85cacc07ba6443387294 9f15c00	no	1
Warnings:					
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9	Specification	CONT2-16247186-SPEC.pdf	190616 adad41d2b332372272ef7567cbb33b6343ccb9e1	no	22
Warnings:					
Information:					
10	Fee Worksheet (SB06)	fee-info.pdf	35133 05e2432359e8c961ce5735909f840f3d6827a473	no	2
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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.

Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

Secrecy Order 37 CFR 5.2:

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

Inventor Information:

Inventor 1

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Steven	Wayne	Goldstein	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	Delray Beach	State/Province	FL	Country of Residence i	US

Mailing Address of Inventor:

Address 1	16193 Bridlewood Circle		
Address 2			
City	Delray Beach	State/Province	FL
Postal Code	33445	Country i	US

Inventor 2

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	Marc	Andre	Boillot	

Residence Information (Select One) ◉ US Residency ○ Non US Residency ○ Active US Military Service

City	Plantation	State/Province	FL	Country of Residence i	US

Mailing Address of Inventor:

Address 1	16193 Bridlewood Circle		
Address 2			
City	Delray Beach	State/Province	FL
Postal Code	33445	Country i	US

Inventor 3

Legal Name

Prefix	Given Name	Middle Name	Family Name	Suffix
	John		Usher	

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Residence Information (Select One)	○ US Residency	◉ Non US Residency	○ Active US Military Service

City	Devon	Country of Residence [i]	GB

Mailing Address of Inventor:

Address 1	16193 Bridlewood Circle		
Address 2			
City	Delray Beach	**State/Province**	FL
Postal Code	33445	**Country** [i]	US

Inventor 4

Legal Name

Prefix	**Given Name**	**Middle Name**	**Family Name**	**Suffix**
	Jason		McIntosh	

Residence Information (Select One)	◉ US Residency	○ Non US Residency	○ Active US Military Service

City	Sugar Hill	**State/Province**	GA	**Country of Residence** [i]	US

Mailing Address of Inventor:

Address 1	16193 Bridlewood Circle		
Address 2			
City	Delray Beach	**State/Province**	FL
Postal Code	33445	**Country** [i]	US

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

Correspondence Information:

Enter either Customer Number or complete the Correspondence Information section below.
For further information see 37 CFR 1.33(a).

☐ **An Address is being provided for the correspondence Information of this application.**

Customer Number	150139		
Email Address	ip@akerman.com	Add Email	Remove Email

Application Information:

Title of the Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION		
Attorney Docket Number	CONT2-16247186	**Small Entity Status Claimed**	☒
Application Type	Nonprovisional		
Subject Matter	Utility		
Total Number of Drawing Sheets (if any)	10	**Suggested Figure for Publication (if any)**	3

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Filing By Reference:

Only complete this section when filing an application by reference under 35 U.S.C. 111(c) and 37 CFR 1.57(a). Do not complete this section if application papers including a specification and any drawings are being filed. Any domestic benefit or foreign priority information must be provided in the appropriate section(s) below (i.e., "Domestic Benefit/National Stage Information" and "Foreign Priority Information").

For the purposes of a filing date under 37 CFR 1.53(b), the description and any drawings of the present application are replaced by this reference to the previously filed application, subject to conditions and requirements of 37 CFR 1.57(a).

Application number of the previously filed application	Filing date (YYYY-MM-DD)	Intellectual Property Authority or Country

Publication Information:

☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)

☐ **Request Not to Publish.** I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application **has not and will not** be the subject of an application filed in another country, or under a multilateral international agreement, that requires publication at eighteen months after filing.

Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Either enter Customer Number or complete the Representative Name section below. If both sections are completed the customer Number will be used for the Representative Information during processing.

Please Select One:	◉ Customer Number	○ US Patent Practitioner	○ Limited Recognition (37 CFR 11.9)
Customer Number	150139		

Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
Registration Number					
Prefix	**Given Name**	Middle Name	**Family Name**	Suffix	Remove
Registration Number					

Additional Representative Information blocks may be generated within this form by selecting the **Add** button.

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Domestic Benefit/National Stage Information:

This section allows for the applicant to either claim benefit under 35 U.S.C. 119(e), 120, 121, 365(c), or 386(c) or indicate National Stage entry from a PCT application. Providing benefit claim information in the Application Data Sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78.
When referring to the current application, please leave the "Application Number" field blank.

Prior Application Status	Pending	Remove

Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
	Continuation in part of	16247186	2019-01-14

Prior Application Status	Patented	Remove

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
16247186	Continuation of	13956767	2013-08-01	10182289	2019-01-15

Prior Application Status	Patented	Remove

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
13956767	Continuation of	12170171	2008-07-09	8526645	2013-09-03

Prior Application Status	Patented	Remove

Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)	Patent Number	Issue Date (YYYY-MM-DD)
12170171	Continuation in part of	12115349	2008-05-05	8081780	2011-12-20

Prior Application Status	Expired	Remove

Application Number	Continuity Type	Prior Application Number	Filing or 371(c) Date (YYYY-MM-DD)
12115349	Claims benefit of provisional	60916271	2007-05-04

Additional Domestic Benefit/National Stage Data may be generated within this form by selecting the **Add** button.

Foreign Priority Information:

This section allows for the applicant to claim priority to a foreign application. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55. When priority is claimed to a foreign application that is eligible for retrieval under the priority document exchange program (PDX) the information will be used by the Office to automatically attempt retrieval pursuant to 37 CFR 1.55(i)(1) and (2). Under the PDX program, applicant bears the ultimate responsibility for ensuring that a copy of the foreign application is received by the Office from the participating foreign intellectual property office, or a certified copy of the foreign priority application is filed, within the time period specified in 37 CFR 1.55(g)(1).

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Application Number	Country[i]	Filing Date (YYYY-MM-DD)	Access Code[i] (if applicable)

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button.

Statement under 37 CFR 1.55 or 1.78 for AIA (First Inventor to File) Transition Applications

☐ This application (1) claims priority to or the benefit of an application filed before March 16, 2013 and (2) also contains, or contained at any time, a claim to a claimed invention that has an effective filing date on or after March 16, 2013.
NOTE: By providing this statement under 37 CFR 1.55 or 1.78, this application, with a filing date on or after March 16, 2013, will be examined under the first inventor to file provisions of the AIA.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Authorization or Opt-Out of Authorization to Permit Access:

When this Application Data Sheet is properly signed and filed with the application, applicant has provided written authority to permit a participating foreign intellectual property (IP) office access to the instant application-as-filed (see paragraph A in subsection 1 below) and the European Patent Office (EPO) access to any search results from the instant application (see paragraph B in subsection 1 below).

Should applicant choose not to provide an authorization identified in subsection 1 below, applicant **must opt-out** of the authorization by checking the corresponding box A or B or both in subsection 2 below.

NOTE: This section of the Application Data Sheet is **ONLY** reviewed and processed with the **INITIAL** filing of an application. After the initial filing of an application, an Application Data Sheet cannot be used to provide or rescind authorization for access by a foreign IP office(s). Instead, Form PTO/SB/39 or PTO/SB/69 must be used as appropriate.

1. Authorization to Permit Access by a Foreign Intellectual Property Office(s)

A. Priority Document Exchange (PDX) - Unless box A in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the European Patent Office (EPO), the Japan Patent Office (JPO), the Korean Intellectual Property Office (KIPO), the State Intellectual Property Office of the People's Republic of China (SIPO), the World Intellectual Property Organization (WIPO), and any other foreign intellectual property office participating with the USPTO in a bilateral or multilateral priority document exchange agreement in which a foreign application claiming priority to the instant patent application is filed, access to: (1) the instant patent application-as-filed and its related bibliographic data, (2) any foreign or domestic application to which priority or benefit is claimed by the instant application and its related bibliographic data, and (3) the date of filing of this Authorization. See 37 CFR 1.14(h)(1).

B. Search Results from U.S. Application to EPO - Unless box B in subsection 2 (opt-out of authorization) is checked, the undersigned hereby **grants the USPTO authority** to provide the EPO access to the bibliographic data and search results from the instant patent application when a European patent application claiming priority to the instant patent application is filed. See 37 CFR 1.14(h)(2).

The applicant is reminded that the EPO's Rule 141(1) EPC (European Patent Convention) requires applicants to submit a copy of search results from the instant application without delay in a European patent application that claims priority to the instant application.

2. Opt-Out of Authorizations to Permit Access by a Foreign Intellectual Property Office(s)

☐ A. Applicant **DOES NOT** authorize the USPTO to permit a participating foreign IP office access to the instant application-as-filed. If this box is checked, the USPTO will not be providing a participating foreign IP office with any documents and information identified in subsection 1A above.

☐ B. Applicant **DOES NOT** authorize the USPTO to transmit to the EPO any search results from the instant patent application. If this box is checked, the USPTO will not be providing the EPO with search results from the instant application.

NOTE: Once the application has published or is otherwise publicly available, the USPTO may provide access to the application in accordance with 37 CFR 1.14.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Applicant Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Applicant 1

If the applicant is the inventor (or the remaining joint inventor or inventors under 37 CFR 1.45), this section should not be completed. The information to be provided in this section is the name and address of the legal representative who is the applicant under 37 CFR 1.43; or the name and address of the assignee, person to whom the inventor is under an obligation to assign the invention, or person who otherwise shows sufficient proprietary interest in the matter who is the applicant under 37 CFR 1.46. If the applicant is an applicant under 37 CFR 1.46 (assignee, person to whom the inventor is obligated to assign, or person who otherwise shows sufficient proprietary interest) together with one or more joint inventors, then the joint inventor or inventors who are also the applicant should be identified in this section.

Clear

◯ Assignee	◯ Legal Representative under 35 U.S.C. 117	◯ Joint Inventor
◉ Person to whom the inventor is obligated to assign.	◯ Person who shows sufficient proprietary interest	

If applicant is the legal representative, indicate the authority to file the patent application, the inventor is:

Name of the Deceased or Legally Incapacitated Inventor:	
If the Applicant is an Organization check here.	☒
Organization Name	Staton Techiya LLC

Mailing Address Information For Applicant:

Address 1	2500 NE 35th Street		
Address 2			
City	Lighthouse Point	**State/Province**	FL
Country[i]	US	Postal Code	33064
Phone Number		Fax Number	
Email Address	Daniel@techiya.co		

Additional Applicant Data may be generated within this form by selecting the Add button.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Assignee Information including Non-Applicant Assignee Information:

Providing assignment information in this section does not substitute for compliance with any requirement of part 3 of Title 37 of CFR to have an assignment recorded by the Office.

Assignee 1

Complete this section if assignee information, including non-applicant assignee information, is desired to be included on the patent application publication. An assignee-applicant identified in the "Applicant Information" section will appear on the patent application publication as an applicant. For an assignee-applicant, complete this section only if identification as an assignee is also desired on the patent application publication.

If the Assignee or Non-Applicant Assignee is an Organization check here. ☒

Organization Name	Staton Techiya LLC

Mailing Address Information For Assignee including Non-Applicant Assignee:

Address 1	2500 NE 35th Street		
Address 2			
City	Lighthouse Point	**State/Province**	FL
Country i	US	Postal Code	33064
Phone Number		Fax Number	
Email Address	Daniel@techiya.co		

Additional Assignee or Non-Applicant Assignee Data may be generated within this form by selecting the Add button.

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Application Data Sheet 37 CFR 1.76	Attorney Docket Number	CONT2-16247186
	Application Number	
Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION	

Signature:

NOTE: This Application Data Sheet must be signed in accordance with 37 CFR 1.33(b). **However, if this Application Data Sheet is submitted with the INITIAL filing of the application and either box A or B is not checked in subsection 2 of the "Authorization or Opt-Out of Authorization to Permit Access" section, then this form must also be signed in accordance with 37 CFR 1.14(c).**

This Application Data Sheet **must** be signed by a patent practitioner if one or more of the applicants is a **juristic entity** (e. g., corporation or association). If the applicant is two or more joint inventors, this form must be signed by a patent practitioner, **all** joint inventors who are the applicant, or one or more joint inventor-applicants who have been given power of attorney (e.g., see USPTO Form PTO/AIA/81) on behalf of **all** joint inventor-applicants.

See 37 CFR 1.4(d) for the manner of making signatures and certifications.

Signature	/John Keady/			Date (YYYY-MM-DD)	
First Name	John	Last Name	Keady	Registration Number	56389

Additional Signature may be generated within this form by selecting the Add button.

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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

As the below named inventor, I hereby declare that:

This declaration is directed to:

☐ The attached application, or

■ United States application or PCT international application number 13/956,767 filed on August 1, 2013.

The above-identified application was made or authorized to be made by me.

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

WARNING:

Petitioner/applicant is cautioned to avoid submitting personal information in documents filed in a patent application that may contribute to identity theft. Personal information such as social security numbers, bank account numbers, or credit card numbers (other than a check or credit card authorization form PTO-2038 submitted for payment purposes) is never required by the USPTO to support a petition or an application. If this type of personal information is included in documents submitted to the USPTO, petitioners/applicants should consider redacting such personal information from the documents before submitting them to the USPTO. Petitioner/applicant is advised that the record of a patent application is available to the public after publication of the application (unless a non-publication request in compliance with 37 CFR 1.213(a) is made in the application) or issuance of a patent. Furthermore, the record from an abandoned application may also be available to the public if the application is referenced in a published application or an issued patent (see 37 CFR 1.14). Checks and credit card authorization forms PTO-2038 submitted for payment purposes are not retained in the application file and therefore are not publicly available.

LEGAL NAME OF INVENTOR

Inventor: Jason McIntosh Date (Optional): 11-15-2018

Signature: [signature]

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

This collection of information is required by 35 U.S.C. 115 and 37 CFR 1.63. 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 1.11 and 1.14. This collection is estimated to take 1 minute 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.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

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☐ The attached application, or

☒ United States application or PCT international application number 13/956,767 filed on August 1, 2013.

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LEGAL NAME OF INVENTOR

Inventor: John Usher Date (Optional) :_____

Signature: _____

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

This collection of information is required by 35 U.S.C. 115 and 37 CFR 1.63. 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 1.11 and 1.14. This collection is estimated to take 1 minute 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.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

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☒ United States application or PCT international application number 13/956,767

filed on August 1, 2013.

The above-identified application was made or authorized to be made by me.

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

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LEGAL NAME OF INVENTOR

Inventor: Steven Wayne Goldstein Date (Optional) :

Signature: Steve Goldstein

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

This collection of information is required by 35 U.S.C. 115 and 37 CFR 1.63. 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 1.11 and 1.14. This collection is estimated to take 1 minute 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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DECLARATION (37 CFR 1.63) FOR UTILITY OR DESIGN APPLICATION USING AN APPLICATION DATA SHEET (37 CFR 1.76)

Title of Invention	METHOD AND DEVICE FOR IN EAR CANAL ECHO SUPPRESSION

As the below named inventor, I hereby declare that:

This declaration is directed to:

- ☐ The attached application, or
- ☒ United States application or PCT international application number 13/956,767 filed on August 1, 2013

The above-identified application was made or authorized to be made by me.

I believe that I am the original inventor or an original joint inventor of a claimed invention in the application.

I hereby acknowledge that any willful false statement made in this declaration is punishable under 18 U.S.C. 1001 by fine or imprisonment of not more than five (5) years, or both.

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LEGAL NAME OF INVENTOR

Inventor: Marc Andre Boillot Date (Optional): 11/12/14

Signature: [signature]

Note: An application data sheet (PTO/SB/14 or equivalent), including naming the entire inventive entity, must accompany this form or must have been previously filed. Use an additional PTO/AIA/01 form for each additional inventor.

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ABSTRACT

An earpiece (100) and acoustic management module (300) for in-ear canal echo suppression control suitable is provided. The earpiece can include an Ambient Sound Microphone (111) to capture ambient sound, an Ear Canal Receiver (125) to deliver audio content to an ear canal, an Ear Canal Microphone (123) configured to capture internal sound, and a processor (121) to generate a voice activity level (622) and suppress an echo of spoken voice in the electronic internal signal, and mix an electronic ambient signal with an electronic internal signal in a ratio dependent on the voice activity level and a background noise level to produce a mixed signal (323) that is delivered to the ear canal (131).

CLAIMS

We claim:

1. An earphone configured to use multiple microphones for noise control, comprising:

a first ambient microphone configured to generate a first ambient signal;

a second ambient microphone configured to generate a second ambient signal;

an ear canal microphone configured to generate an ear canal signal;

a speaker;

an eartip, configured to seal the earphone between a first side and a second side of the earphone;

a memory that stores instructions; and

a processor, wherein the processor is operatively connected to the ambient microphone, wherein the processor is operatively connected to the ear canal microphone, wherein the processor is operatively connected to the speaker, wherein the processor is operatively connected to the memory, wherein the processor is configured to execute the instructions to perform operations, the operations comprising:

receiving a first microphone signal, wherein the first microphone signal is at least one of the first ambient signal or a modified first ambient signal;

receiving a second microphone signal, wherein the second microphone signal is at least one of the ear canal signal or a modified ear canal signal

generating a background noise reduction signal using the first and second microphone signals; and

sending the background noise reduction signal to the speaker.

2. The earphone according to claim 1, wherein the operations further comprise:

sending an audio content signal to the speaker.

3. The earphone according to claim 1, wherein the operations further comprise:

generating an ambient signal from at least one of the first ambient signal or the second ambient signal or a combination of both.

4. The earphone according to claim 3, wherein the operations further comprise:

sending a modified ambient signal to the speaker.

5. The earphone according to claim 4, where the modified ambient signal is the ambient signal.

6. The earphone according to claim 4, wherein the operations further comprise:

generating the modified ambient signal by filtering the ambient signal.

7. The earphone according to claim 3, wherein the operations further comprise:

sending a modified audio content signal to the speaker.

8. The earphone according to claim 7, wherein the operations further comprise:

generating the modified audio content signal by combining an audio content signal with the modified ambient signal.

9. The earphone according to claim 4, wherein the operations further comprise:

generating a modified ambient signal by filtering the ambient signal and modifying an amplitude of at least one frequency of the ambient signal.

10. The earphone according to claim 8, wherein the step of combining includes combining the audio signal and the modified ambient signal at a predetermined ratio value.

11. The earphone according to claim 10, where the ratio value is multiplied by the modified ambient signal prior to combining with the audio signal.

12. The earphone according to claim 3, wherein the operations further comprise:

analyzing at least one of the ambient signal or the ear canal signal or a combination of both, to detect a user's voice.

13. The earphone according to claim 12, wherein the operations further comprise:

analyzing at least one of the ambient signal or the ear canal signal or a combination of both, to generate a voice activity level.

14. The earphone according to claim 13, wherein the operations further comprise:

sending an audio content signal to the speaker if the voice activity level is below a predetermined threshold.

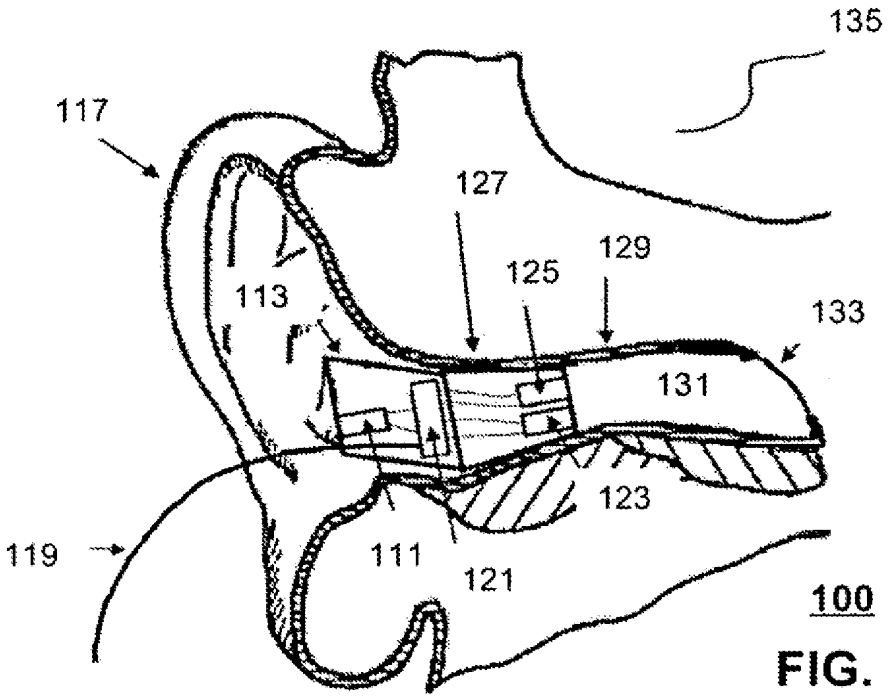

135
117
127
129
125
113
133
131
123
119
111
121
100

FIG. 1

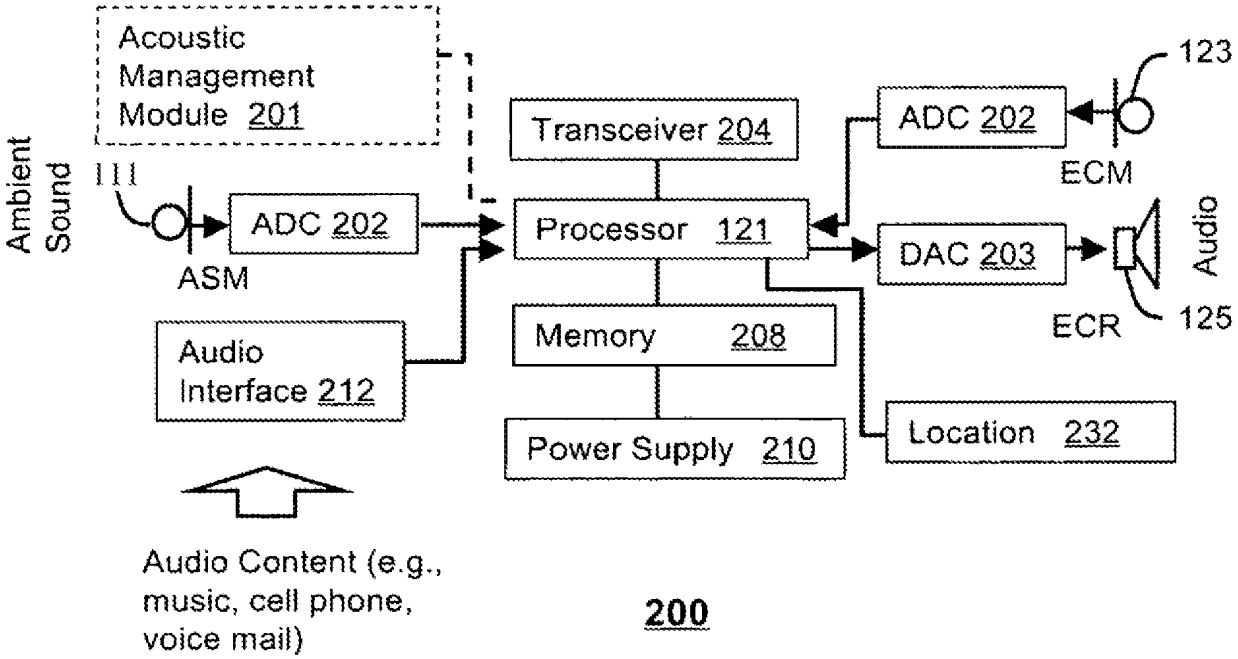

Acoustic Management Module 201
Ambient Sound
111
ADC 202
ASM
Transceiver 204
Processor 121
Memory 208
Power Supply 210
Audio Interface 212
ADC 202
123
ECM
DAC 203
Audio
ECR
125
Location 232
Audio Content (e.g., music, cell phone, voice mail)
200

FIG. 2

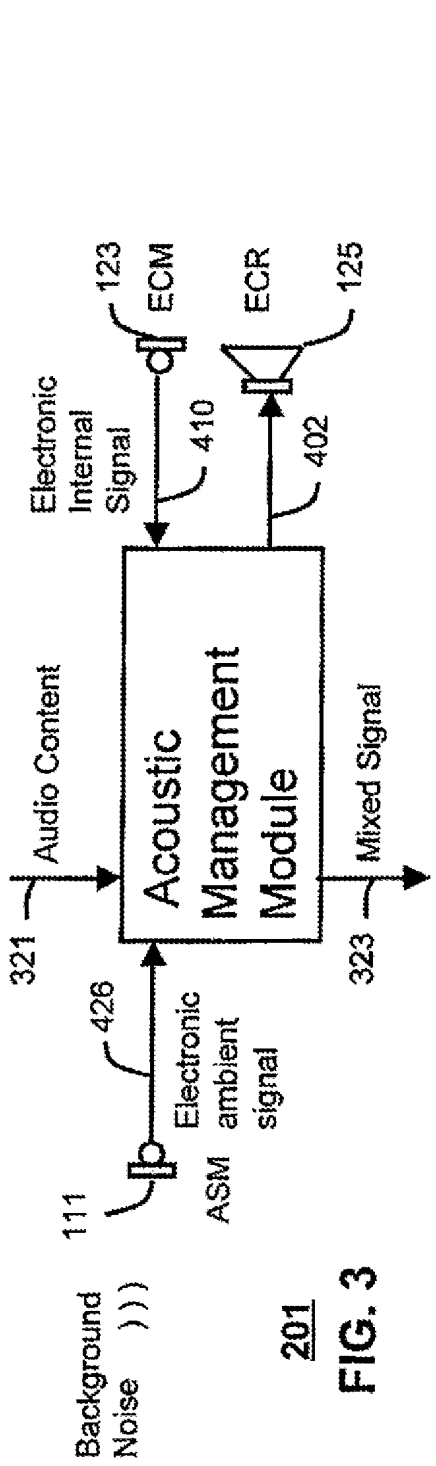

Background Noise
111
ASM
426
Electronic ambient signal
321
Audio Content
Acoustic Management Module
323
Mixed Signal
Electronic Internal Signal
410
123
ECM
ECR
125
402
201

FIG. 3

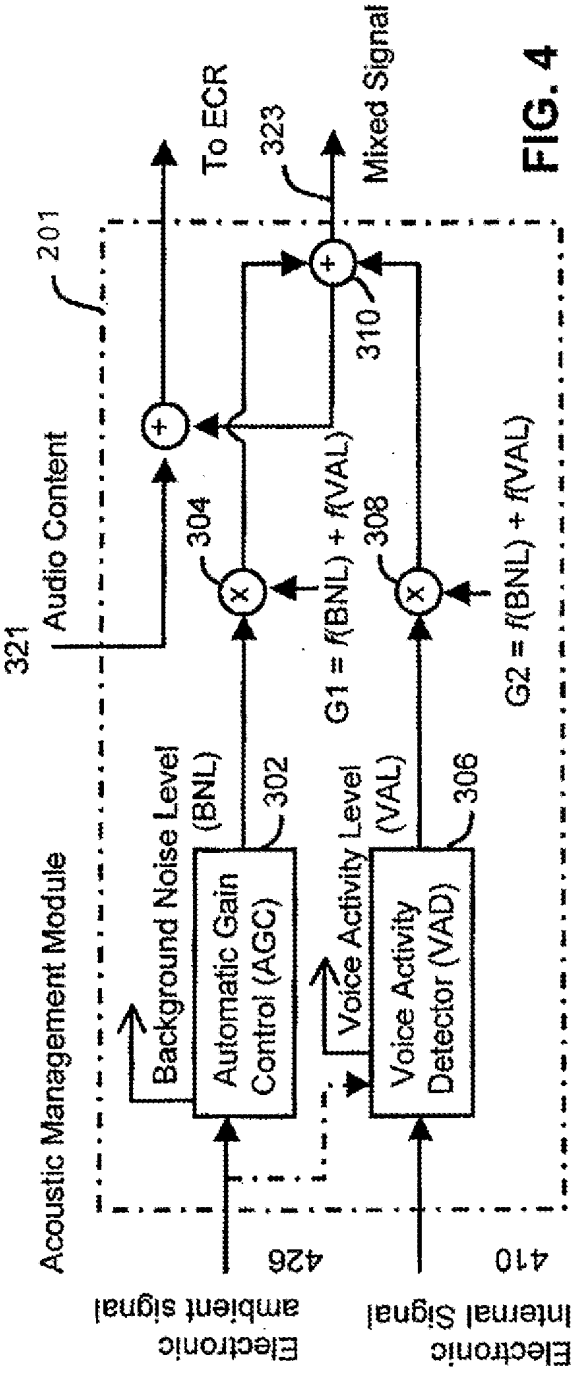

Acoustic Management Module
321
Audio Content
201
To ECR
323
Mixed Signal
Background Noise Level (BNL)
Automatic Gain Control (AGC)
302
304
Voice Activity Level (VAL)
Voice Activity Detector (VAD)
306
308
310
G1 = f(BNL) + f(VAL)
G2 = f(BNL) + f(VAL)
Electronic ambient signal
426
Electronic Internal Signal
410

FIG. 4

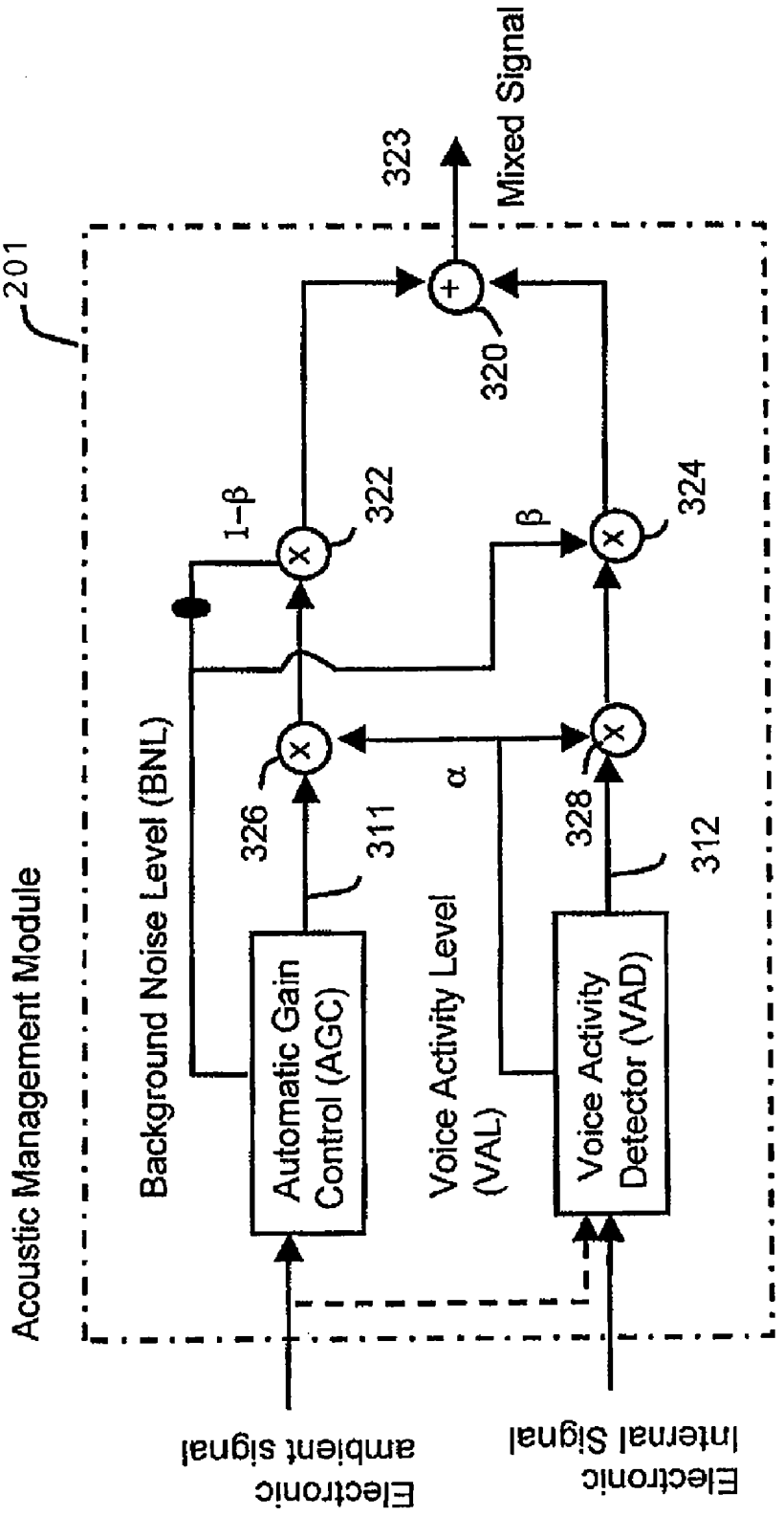
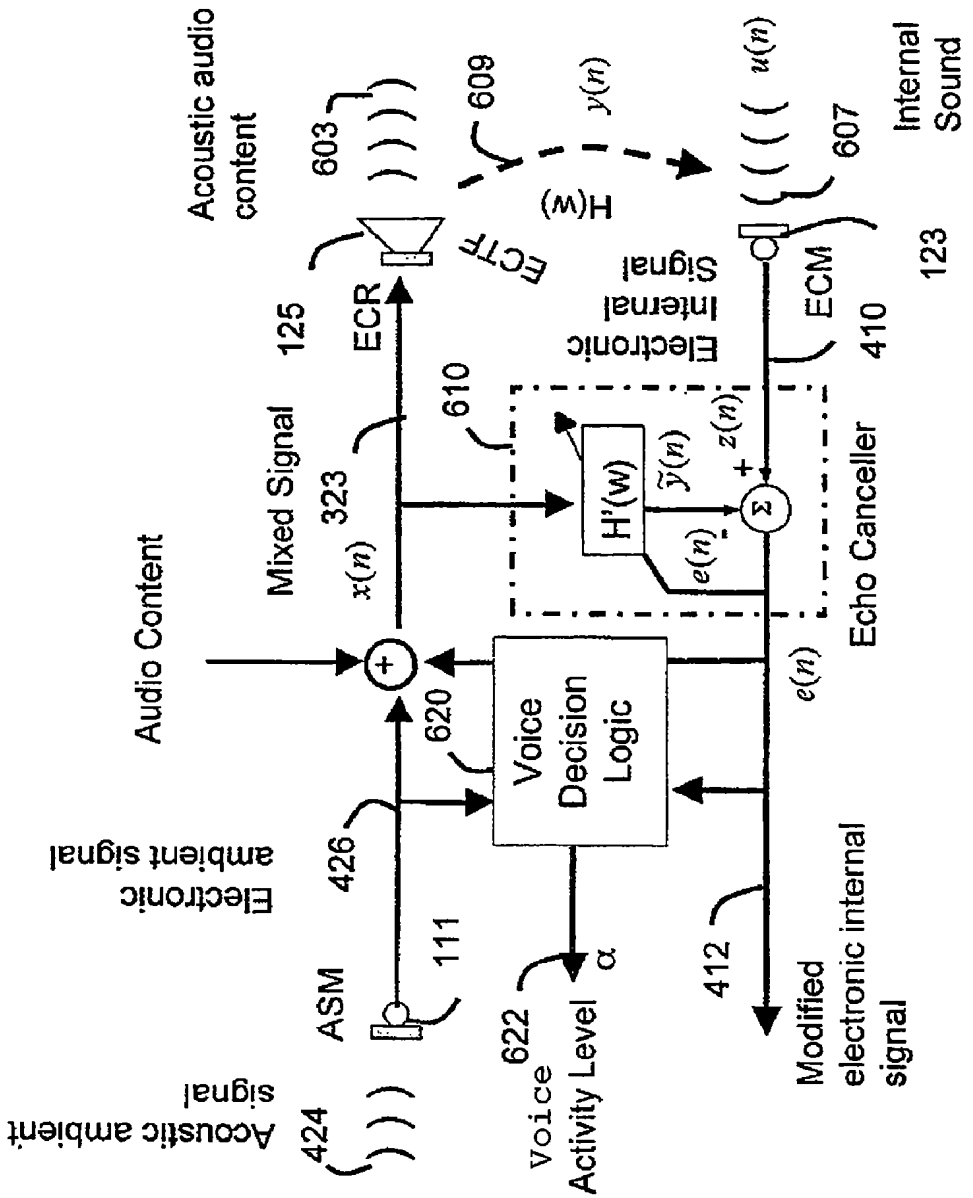

Acoustic Management Module
201
Electronic ambient signal
Electronic Internal Signal
Automatic Gain Control (AGC)
Voice Activity Detector (VAD)
Background Noise Level (BNL)
Voice Activity Level (VAL)
311
312
326
328
α
1–β
β
322
324
320
323
Mixed Signal

FIG. 5


Acoustic audio content
603
609
$y(n)$
H(w)
$u(n)$
607
Internal Sound
123
ECTT
ECR
125
Electronic Internal Signal
ECM
410
Mixed Signal
323
$x(n)$
610
H'(w)
$\tilde{y}(n)$
$z(n)$
$e(n)$
Audio Content
620
Voice Decision Logic
$e(n)$
Echo Canceller
Electronic ambient signal
426
ASM
111
412
Modified electronic internal signal
Acoustic ambient signal
424
Voice Activity Level
622
α

600

FIG. 6

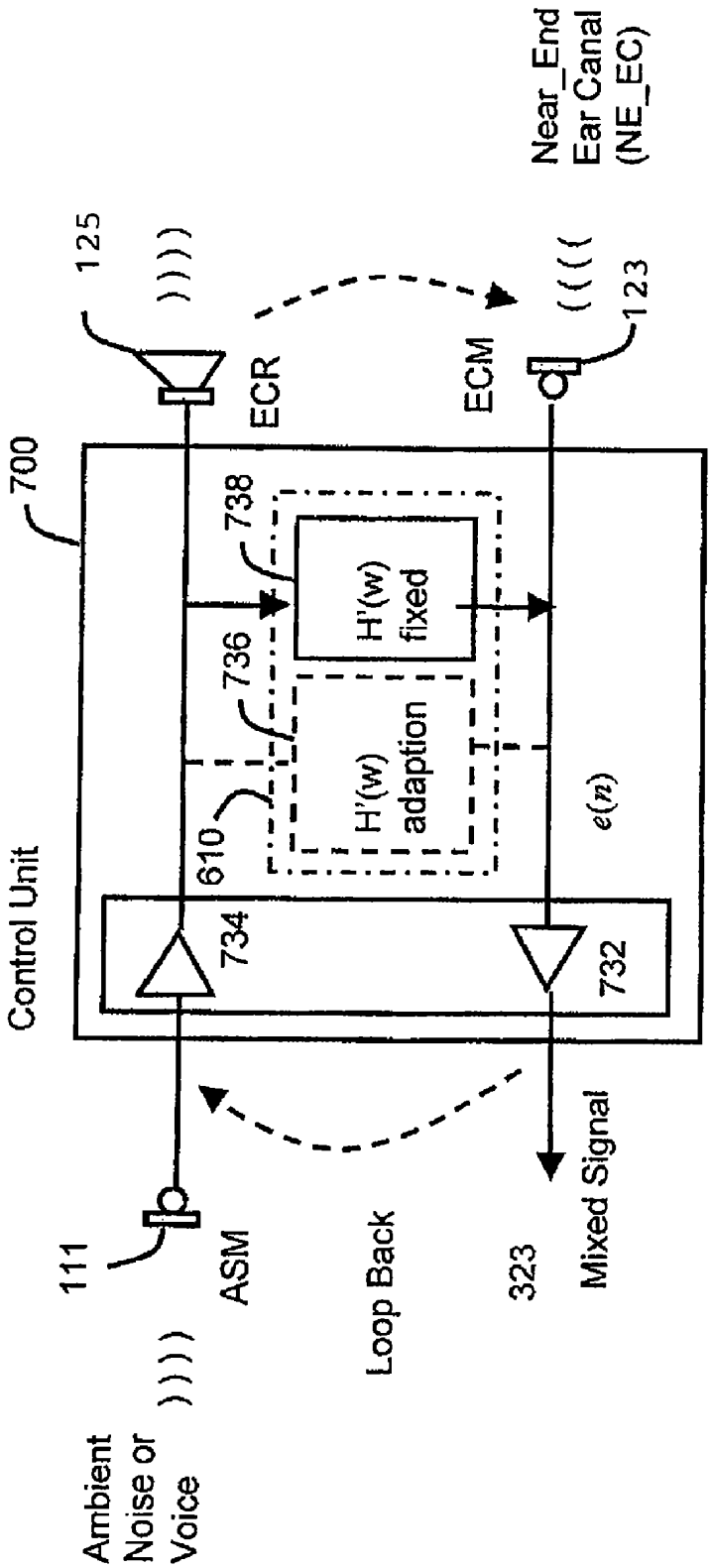

Near_End
Ear Canal
(NE_EC)
125
123
ECR
ECM
700
738
H'(w)
fixed
736
H'(w)
adaption
610
e(n)
Control Unit
734
732
111
ASM
Loop Back
323
Mixed Signal
Ambient
Noise or
Voice

FIG. 7

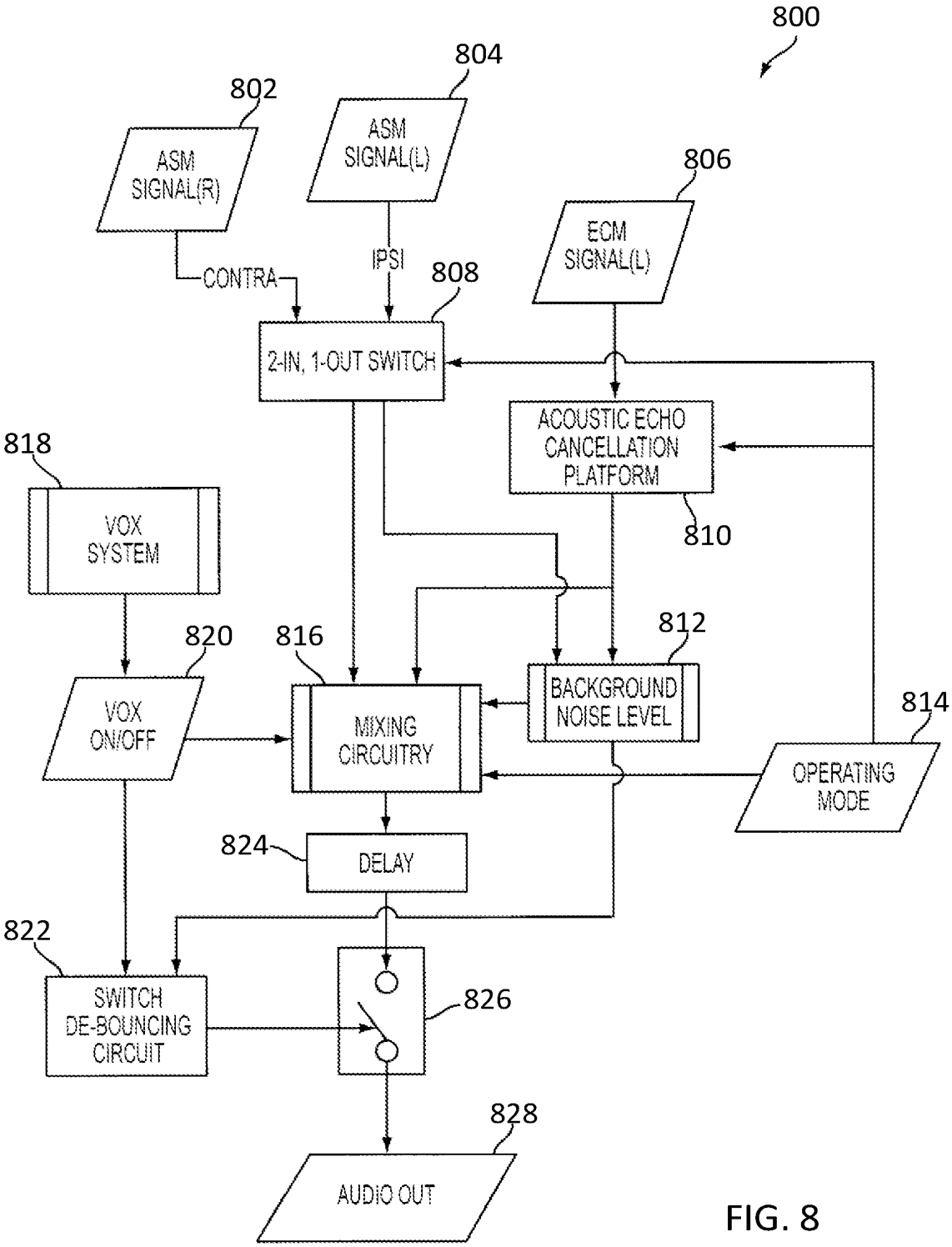

800
804
802
ASM SIGNAL(L)
ASM SIGNAL(R)
806
ECM SIGNAL(L)
IPSI
CONTRA
808
2-IN, 1-OUT SWITCH
ACOUSTIC ECHO CANCELLATION PLATFORM
818
VOX SYSTEM
810
820
816
812
VOX ON/OFF
MIXING CIRCUITRY
BACKGROUND NOISE LEVEL
814
OPERATING MODE
824
DELAY
822
SWITCH DE-BOUNCING CIRCUIT
826
828
AUDIO OUT

FIG. 8

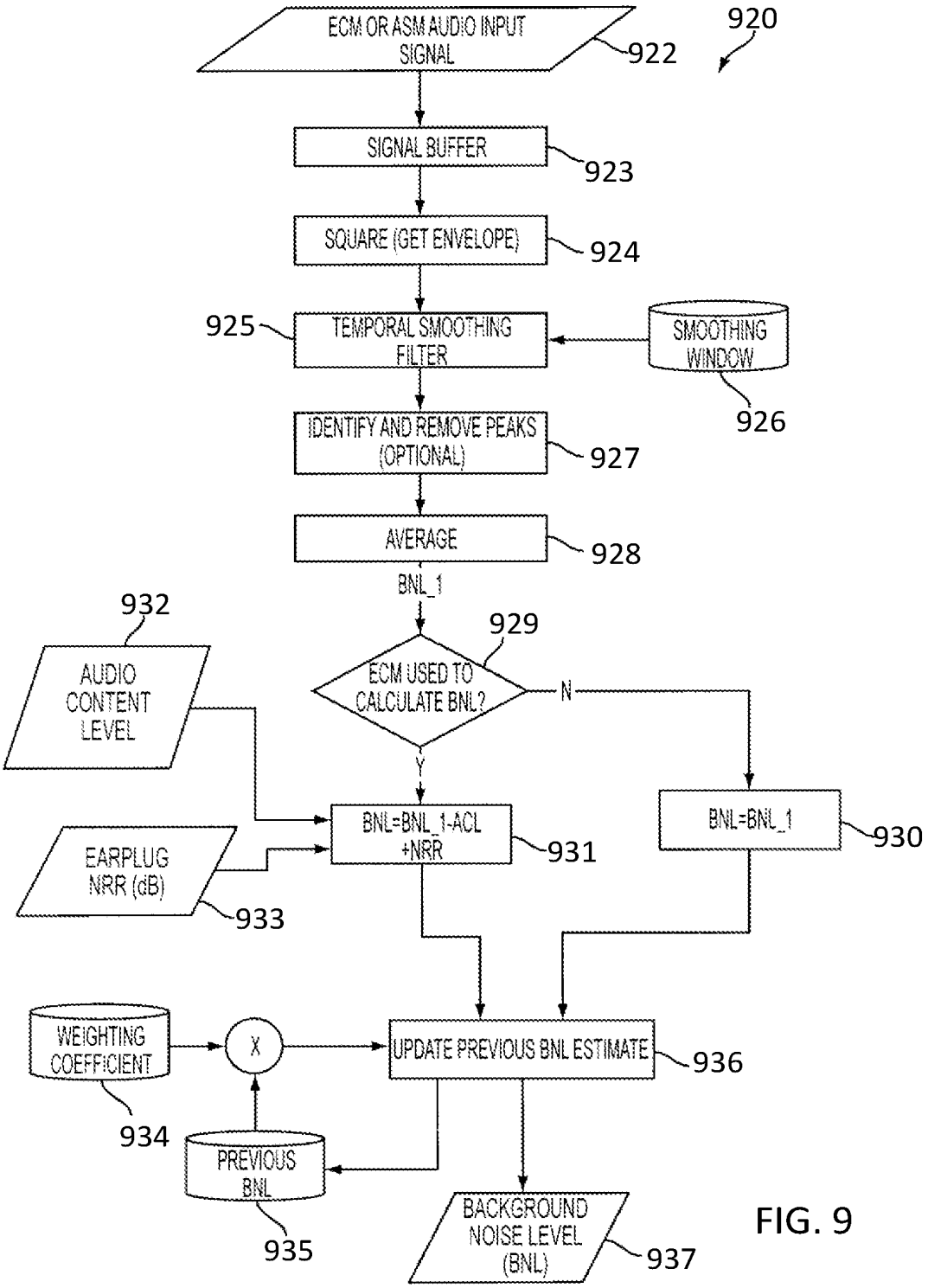

ECM OR ASM AUDIO INPUT SIGNAL
922
920
SIGNAL BUFFER
923
SQUARE (GET ENVELOPE)
924
925
TEMPORAL SMOOTHING FILTER
SMOOTHING WINDOW
926
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
927
AVERAGE
928
BNL_1
932
AUDIO CONTENT LEVEL
929
ECM USED TO CALCULATE BNL?
N
Y
BNL=BNL_1-ACL +NRR
931
BNL=BNL_1
930
EARPLUG NRR (dB)
933
WEIGHTING COEFFICIENT
X
UPDATE PREVIOUS BNL ESTIMATE
936
934
PREVIOUS BNL
935
BACKGROUND NOISE LEVEL (BNL)
937

FIG. 9

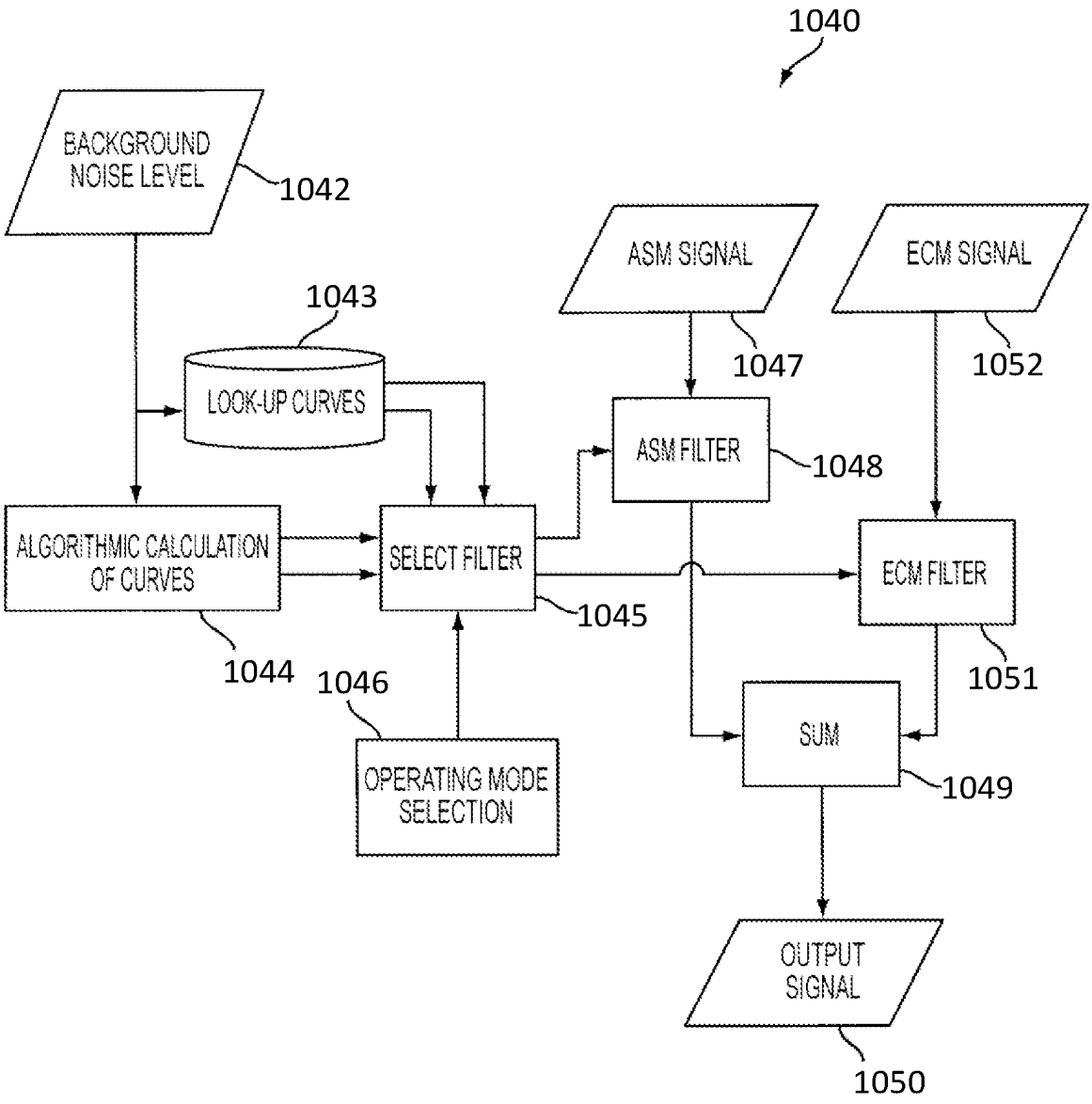

1040
BACKGROUND NOISE LEVEL
1042
1043
LOOK-UP CURVES
ASM SIGNAL
1047
ECM SIGNAL
1052
ASM FILTER
1048
ALGORITHMIC CALCULATION OF CURVES
SELECT FILTER
1045
ECM FILTER
1044
1046
1051
SUM
OPERATING MODE SELECTION
1049
OUTPUT SIGNAL
1050

FIG. 10

FIG. 11

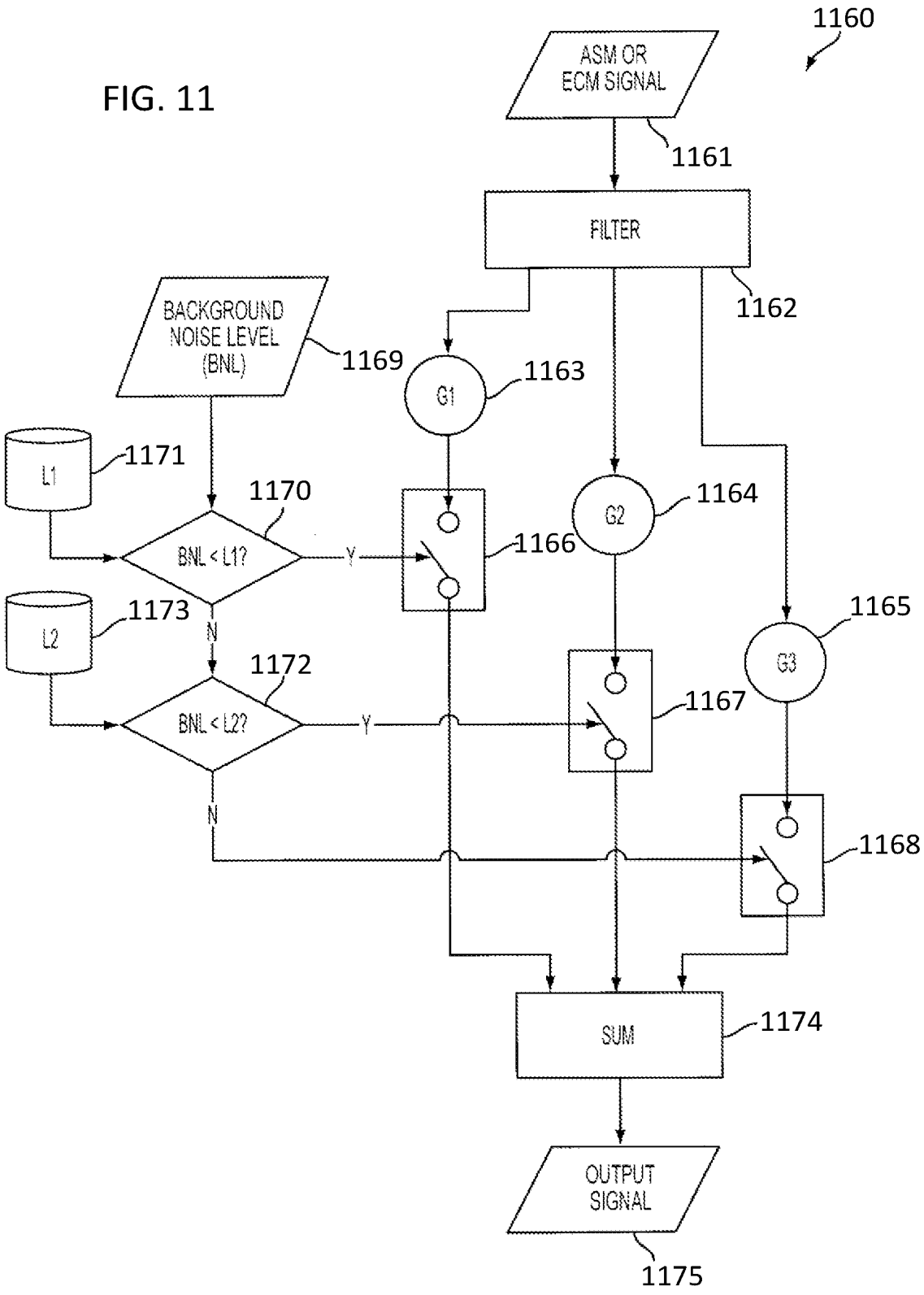
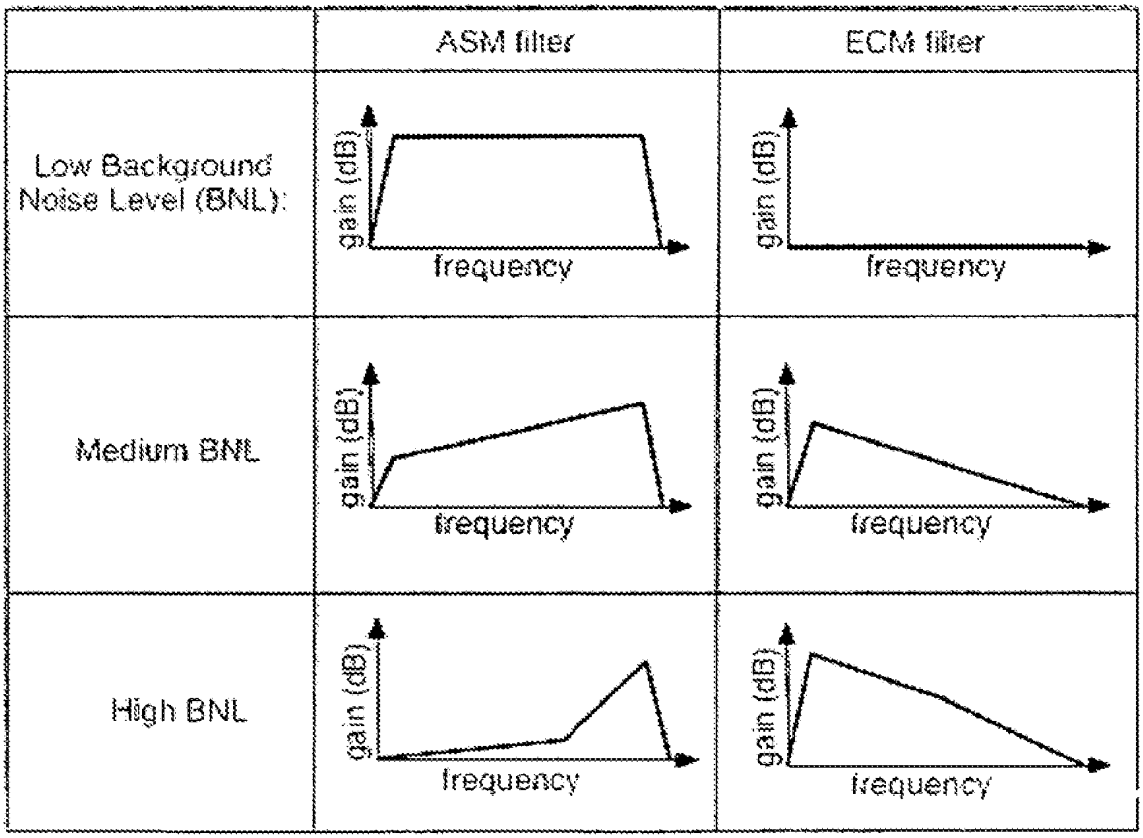

1160
ASM OR ECM SIGNAL
1161
FILTER
1162
BACKGROUND NOISE LEVEL (BNL)
1169
G1
1163
L1
1171
1170
G2
1164
BNL < L1?
Y
1166
L2
1173
N
1165
G3
1172
BNL < L2?
Y
1167
N
1168
SUM
1174
OUTPUT SIGNAL
1175

FIG. 12


ASM filter
ECM filter
Low Background Noise Level (BNL):
Medium BNL
High BNL
gain (dB)
frequency