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(54) Noise canceling microphone array

(57) A method for enhancing discrimination of sound received from a sound source (54) relative to acoustic interference, including: providing a plurality of sound sensors (52) in predetermined positions, and receiving respective signals from the plurality of sound sensors responsive to the interference and to the sound source.

Respective characteristics of the plurality of signals in each of a plurality of spectral bands are determined, and the characteristics are analyzed to compute a spectral gain function (132) which discriminately enhances a portion of the signals that is associated with the sound source. The signals from one or more of the plurality of sensors are processed to generate a combined master signal, and the spectral gain function is applied to the master signal so as to generate an output signal in which the portion of the signals associated with the sound source is enhanced relative to that due to the acoustic interference.

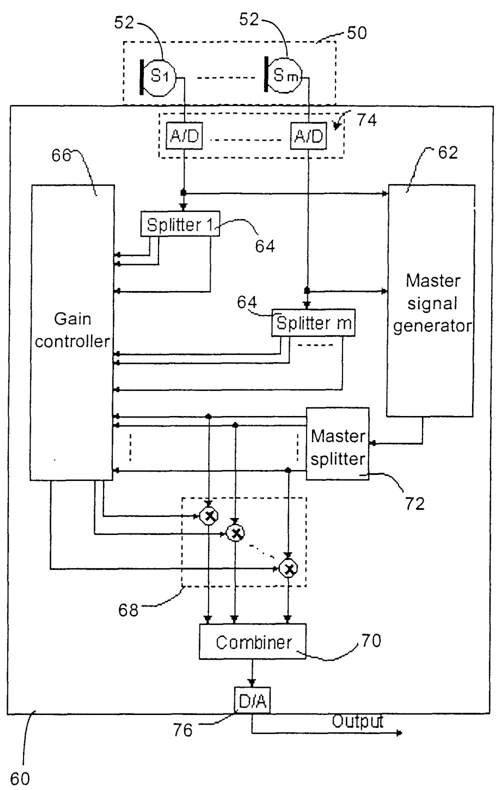

Fig. 4

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Description

FIELD OF THE INVENTION

[0001] This invention relates generally to the field of microphones, and specifically to noise cancellation and signal enhancement for microphones.

BACKGROUND TO THE INVENTION

[0002] Noise, echoes, and other interference may significantly degrade the perceptual quality of signals recorded by conventional microphones. Furthermore, if a noise-contaminated signal is passed to a voice encoder for compression, the quality of the decompressed signal may suffer further due to the encoder's inability to separate the noise from the signal. Acoustical echoes picked up by a microphone, arising for example, in speakerphones, hands-free communications, or teleconferencing, may be very annoying on the far side of the connection. When an echo is present, the only solution for truly full-duplex communications is to use acoustical echo canceling. U.S. Patent 5,305,307 to Chu, and U.S. Patent 5,566,167 to Duttweiler, which are incorporated herein by reference, describe methods of acoustical echo cancellation.

[0003] Fully-effective echo canceling, however, is possible only in a digital domain, and it is computationally and memory-expensive. Even good acoustic echo canceling may be very difficult to achieve in real acoustic environments. The problem is further complicated when cheap or miniature loudspeakers are used, due to possible, sometimes artificially introduced, nonlinearity of the loudspeaker characteristics. When voice recognition software is used, additional interference (for example, voices of people or other sound sources other than the main speaker) may be a serious problem and thus needs to be avoided.

[0004] There is a demand for an acquisition device or technique that either picks up or somehow extracts a sound of interest from among other sounds. Alternatively, such a device or technique may be described as suppressing interference (noise, echoes, other speakers, etc.) in an acquired signal and thus improving the signal-to-noise ratio (SNR). To achieve these goals, the sound of interest must be in some way physically different from the unwanted interference. In the current state of the art, there are three approaches, based on different physical principles, for improving signal-to-noise ratio. These approaches may be applied separately or in combination.

[0005] The first approach utilizes differences in statistical properties between a signal and noise. U.S. Patent 4,185,168, to Graupe et al., and U.S. Patent 5,768,473, to Eatwell et al., which are incorporated he::ein by reference, use this approach. The noise is assumed to have stable, "near-stationary" characteristics compared to the signal. The term near-stationary means that the noise spectrum changes relatively slowly with respect to the spectrum of the signal. It is then possible to estimate the power levels of the noise in different frequency bands. Simultaneously, the short-term power levels of the signal in the same frequency bands are monitored. One or more frequency bands in the output signal are then suppressed or enhanced depending on the current signal-to-noise ratio.

[0006] If only one band is used, then this approach reduces to noise gating and is generally done in the analog domain. This is illustrated in U.S. Patent 5,838,269, to Xie, which is incorporated herein by reference. An obvious shortcoming of this approach is its inability to remove transient noise or noise whose spectral characteristic varies in a similar manner to that of the signal. It is hence useless for removing echoes, reverberations and voice interference. With a high noise level, this approach may also distort the signal when the signal-to-noise ratio in certain frequency bands is close to unity.

[0007] Fig. 1 illustrates a second approach to improving signal-to-noise ratio, using a noise-canceling microphone. The figure shows an acoustic noise-canceling microphone 20 and an electronic noise-canceling microphone 30. Noise canceling microphones have two openings, one close to the sound source (the front) and one farther from the source (the rear). Noise-canceling microphones utilize a net pressure difference caused by the separation of the openings. In an acoustic noise-canceling microphone, a single diaphragm 22 is displaced by the net pressure difference, and such displacement is reflected in the output signal. In an electronic noise-canceling microphone, two bundled microphones 32 and 34 are used, and the difference is computed electronically.

[0008] Noise-canceling microphones operate on the assumption that the noise affects both openings substantially identically, so that the net pressure difference generated by the noise is effectively zero. Conversely, the signal generates a non-zero net pressure difference. For a sound wave traveling from the front to the rear, the net pressure difference is affected by both the phase difference and the sound pressure (the amplitude) difference along the wave. For a source generating a generally spherical sound wave, the sound pressure is inversely proportional to the distance to the sound source. The sensitivity of the microphone is thus a function of the separation of the openings, and to increase the sensitivity, the separation needs to be relatively large. On the other hand, the larger the separation, the larger the differences in phase shift over the microphone operating frequency range. Due to the phase restrictions, the separation between the openings is typically restricted to be less than 15 mm. Such a small separation only provides enough sound pressure difference when the sound source is very close to the microphone, i.e., when the microphone-source distance is less than about 3 cm. Hence using noise-canceling microphones is mainly restricted to headsets or other devices mounted or held very close to a speaker's mouth.

[0009] A third approach to coping with noise and interfering signals consists of using superdirectional microphones. Such microphones attempt to receive and amplify sounds coming from a relatively narrow range of angles about a directional axis intercepting the sound source of interest. Superdirectional microphones may also be built either acoustically or electronically. In the latter case, they are generally called "microphone arrays."

[0010] Fig. 2 is a schematic diagram of a microphone array system 36 comprising a microphone array 38 and a processor 42. Array 38 consists of two or more individual sound pressure sensors 40 distributed along an axis. Each individual sensor may be either omnidirectional, i.e., having a gain substantially fixed regardless of the direction of the incoming signal, or unidirectional, wherein the gain is a function of the direction of the incoming signal. Processor 42 combines input signals from each individual sensor so as to generate an output signal that discriminates between the sounds in the direction of interest and sounds from other directions. To achieve the discrimination, the processor computes the output signal as a linear combination of the input signals. In some cases, the input signals are filtered so that processor 42 may better exploit phase differences between individual sensors 40. Such phase differences are caused by the spatial separation of the individual sensors and by the angular separation between the sound source of interest and interfering sounds.

[0011] A microphone array system may be either fixed or adaptive, depending on the type of processing provided. In a fixed microphone array system, individual filters associated with the microphones are fixed, and do not depend on the signals acquired by individual sensors. The filters are chosen so that the array receives signals from a direction of interest, and attenuates all signals arriving from directions other than the direction of interest. In an adaptive array system, the filters are automatically adapted during array operation, so as to better deal with varying specific situations.

[0012] The Audio and Hi-Fi Handbook (2nd Edition, 1995), by Ian R. Sinclair (Ed), published by Butterworth-Heinemann, and U.S. Patent 5,825,898, to Marash, which is incorporated herein by reference, describe superdirectional microphones. These microphones operate under the far field assumption, meaning that the distance between the microphone array and the sound or interference source is assumed to be substantially larger than the dimensions of the array itself. In this case an acoustic wave approaching the microphone may be regarded as a plane wave. The distance must typically be substantially larger than 100 cm.

[0013] Thus, using superdirectional microphones has two major limitations. First, the use is limited to situations where there is a relatively large distance between the microphone array and the sound source. Second, the microphone array is not able to discriminate between the sound source of interest and a source that is closer or farther away but which lies in the same direction. For example, using superdirectional microphones in small or reverberant rooms does not generally provide a significant improvement in signal-to-noise ratio because of multiple wall reflections and because of the diffused character of the sounds in these rooms.

[0014] Thus the prior art approach to noise cancellation does not provide a microphone system or apparatus having noise-canceling characteristics for sources in the middle range of microphone-source distances (distances of the order of 3-100cm), or which are capable of dealing with all types of acoustic interference.

SUMMARY OF THE INVENTION

[0015] It is an object of the present invention to provide improved methods and devices for suppressing acoustic interference received by a microphone.

[0016] It is an object of some aspects of the present invention to provide microphone apparatus having noise-canceling characteristics in a middle range of distances (3-100cm).

[0017] It is a further object of some aspects of the present invention to provide microphone apparatus and methods of signal processing therefor having an improved signal-to-noise ratio irrespective of the nature of interfering signals.

[0018] It is yet a further object of some aspects of the present invention to provide methods and apparatus which are able to discriminate signals lying outside of a given angular or distance range.

[0019] In preferred embodiments of the present invention, microphone array apparatus comprises a set of two or more sound pressure sensors separated in space and a signal processor, which may be either analog or digital. The processor comprises a master signal generator, a set of frequency band splitters, a gain controller and a signal combiner. The master signal generator combines input signals from one or more of the pressure sensors to produce a master signal, having a fixed or adaptive beam pattern, by methods known in the art. Signals from each of the sensors are split into different, predetermined frequency bands by the splitters, and the split-band signals are then fed into the gain controller, which generates a preset or adaptive gain function for each of the different frequency bands. The master signal is split into the same, predetermined frequency bands. The gain function produced by the gain controller is applied to the master signal bands, and an output signal is then reconstructed from the bands by the combiner.

[0020] The gain function generated by the gain controller utilizes instantaneous power and/or phase differences within the different frequency bands from individual sensors and, optionally, from the master signal. Applying the gain function to the master signal enables the array to discriminate between signals coming from different directions and distances. In particular, the array is able to discriminate signals from sources within the range of about 3-100 cm, as distinct from sources outside this range. Correlation between individual frequency bands may also be taken into account by modifying the gain depending on the overall spectral content of the input signals. For example, the powers may be smoothed before being used, and the total output gain may be modified based on how many individual gain functions exceeded some threshold value.

[0021] There is therefore provided, in accordance with a preferred embodiment of the present invention, a method for enhancing discrimination of sound received from a sound source relative to acoustic interference, including:

providing a plurality of sound sensors in predetermined positions;
receiving respective signals from the plurality of sound sensors responsive to the interference and to the sound source;
determining respective characteristics of the plurality of signals in each of a plurality of spectral bands;
analyzing the determined characteristics to compute a spectral gain function which discriminately enhances a portion of the signals that is associated with the sound source;
processing the signals from one or more of the plurality of sensors to generate a combined master signal; and
applying the spectral gain function to the master signal so as to generate an output signal in which the portion of the signals associated with the sound source is enhanced relative to that due to the acoustic interference.

[0022] Preferably, applying the spectral gain function includes splitting the master signal into a plurality of spectral bands corresponding to the plurality of bands with respect to which the characteristics are determined, and applying a gain factor to each of the bands.

[0023] Preferably, analyzing the determined characteristics includes determining a gain function responsive to a power difference of the signals received from the sound sensors.

[0024] Alternatively, analyzing the determined characteristics includes determining a gain function responsive to a phase difference of the signals received from the sound sensors.

[0025] Preferably, processing the one or more signals to generate the master signal includes summing respective spectral components of the one or more signals in at least one frequency band.

[0026] Alternatively, processing the one or more signals to generate the master signal includes combining the signals responsive to relative phases thereof so as to enhance a contribution to the master signal of sound coming from a preferred direction.

[0027] Preferably, receiving the respective signals includes using a Fast Fourier Transform (FFT), and wherein applying the spectral gain function includes using an inverse FFT.

[0028] Preferably, analyzing the determined characteristics includes selecting a sensitivity region within which the sound source is detected.

[0029] Preferably, the sensitivity region includes distances in a range of 3-100 cm from the plurality of sound sensors.

[0030] Preferably, the plurality of sensors includes at least one omnidirectional sensor.

[0031] Alternatively, the plurality of sensors includes at least one unidirectional sensor.

[0032] Preferably, computing the gain function includes computing the function responsive to a unidirectional sensor gain function.

[0033] There is further provided, in accordance with a preferred embodiment of the present invention, a method for enhancing discrimination of sound received from a source in a given location relative to acoustic interference, including:

providing an array of sound sensors in a predetermined position;
receiving respective signals from the array of sound sensors responsive to the interference and to the sound from the source;
analyzing the signals to identify one or more characteristics of sound received from within a selected range of distances that includes the distance of the location of the source from the position of the array;
determining a gain function responsive to the identified characteristics; and
applying the gain function to the received signals so as to generate an output signal in which a portion of the signals corresponding to sound received from within the selected range of distances is enhanced relative to sound from outside the range.

[0034] Preferably, determining the gain function includes determining a gain function responsive to a power difference of the signals received from the sound sensors.

[0035] Preferably, determining the gain function includes determining a gain function responsive to a phase difference of the signals received from the sound sensors.

[0036] Preferably, analyzing the signals includes determining respective characteristics of the signals in each of a plurality of spectral bands.

[0037] Alternatively, determining the gain function includes determining a gain function using at least one of the spectral bands and applying the function to the other bands.

[0038] Preferably, analyzing the signals includes using a Fast Fourier Transform (FFT), and wherein applying the gain function includes using an inverse FFT to generate the output signal.

[0039] Preferably, the array of sensors includes at least one omnidirectional sensor.

[0040] Alternatively, the array of sensors includes at least one unidirectional sensor.

[0041] Preferably, determining the gain function includes computing the function responsive to a unidirectional sensor gain function.

[0042] There is further provided, in accordance with a preferred embodiment of the present invention, apparatus for enhancing discrimination of sound received from a sound source relative to acoustic interference, including:

a plurality of sound sensors which generate a respective plurality of signals responsive to the interference and to the sound source;
a plurality of splitters, which divide the respective plurality of signals into a plurality of spectral bands;
a master signal generator, which generates a master signal responsive to at least one of the plurality of signals;
a gain controller, which computes a spectral gain function which discriminately enhances a portion of the signals that is associated with the sound source responsive to the signals; and
a signal combiner, which applies the spectral gain function to the master signal so as to generate an output signal in which the portion of the signals associated with the sound source is enhanced relative to that due to the acoustic interference.

[0043] Preferably, the master signal generator includes a splitter which splits the master signal into a plurality of spectral bands corresponding to the plurality of bands into which the plurality of splitters divide the signals.

[0044] Preferably, the gain controller computes the gain function responsive to a power difference of the signals received from the sound sensors.

[0045] Alternatively, the gain controller computes the gain function responsive to a phase difference of the signals received from the sound sensors.

[0046] Preferably, the plurality of sound sensors includes at least one omnidirectional sensor.

[0047] Alternatively, the plurality of sound sensors comprises at least one unidirectional sensor.

[0048] Preferably, the gain controller computes the spectral gain function responsive to a unidirectional sensor gain function.

[0049] There is further provided, in accordance with a preferred embodiment of the present invention, apparatus for enhancing discrimination of sound received from a source in a given location relative to acoustic interference, including:

an array of sound sensors in a predetermined position, which sensors generate a respective plurality of signals responsive to the interference and to the sound source;
a gain controller, which analyzes the signals to identify one or more characteristics of sound received from within a selected range of distances that includes the distance of the location of the source from the position of the array, and which determines a gain function responsive to the identified characteristics; and
a signal combiner, which applies the gain function to the received signals so as to generate an output signal in which a portion of the signals corresponding to sound received from within the selected range of distances is enhanced relative to sound from outside the range.

[0050] Preferably, the apparatus includes a plurality of splitters, which respectively split the signals received from the sensors into a plurality of spectral bands.

[0051] Preferably, the apparatus includes a master signal generator, which generates a master signal responsive to at least one of the plurality of signals, and to which master signal the signal combiner applies the gain function.

[0052] Preferably, the array of sound sensors includes at least one omnidirectional sensor.

[0053] Alternatively, the array of sound sensors includes at least one unidirectional sensor.

[0054] Preferably, the gain controller computes the gain function responsive to a unidirectional sensor gain function

[0055] The present invention will be more fully understood from the following detailed description of the preferred embodiments thereof, taken together with the drawings in which:

BRIEF DESCRIPTION OF THE DRAWINGS

[0056]

Fig. 1 shows an acoustic noise-canceling microphone and an electronic noise-canceling microphone, as are known in the art;

Fig. 2 is a schematic diagram of a microphone array system comprising a microphone array and a processor, as is known in the art;

Fig. 3, which shows a schematic block diagram of a microphone array system, according to a preferred embodiment of the present invention;

Fig. 4 is a schematic block diagram illustrating the operation of a processor in the system of Fig. 3, in accordance with a preferred embodiment of the present invention;

Fig. 5 is a schematic block diagram illustrating the operation of an alternative processor for the system of Fig. 3, in accordance with a preferred embodiment of the present invention;

Fig. 6 is a schematic layout of a microphone array, illustrating a method for calculating a gain function for the array, according to a preferred embodiment of the present invention;

Fig. 7 is a graphical plot of acoustic sensitivity for the array of Fig. 6, according to a preferred embodiment of the present invention; and

Fig. 8 is a graph showing theoretical and actual gain functions for the array of Fig. 6, according to a preferred embodiment of the present invention.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

[0057] Reference is now made to Fig. 3, which is a schematic block diagram of a microphone array system 48, according to a preferred embodiment of the present invention. System 48 comprises an array 50 of sound pressure sensors 52 and a processor 60, which processes signals received from the sensors in order to reduce the effect of noise or other interference present in the signals from the sensors. Preferably, pressure sensors 52 are omnidirectional microphones having substantially similar characteristics and operating over the complete band of audible sound frequencies. Alternatively, pressure sensors 52 are unidirectional microphones.

[0058] Most preferably, distances between sensors 52 are chosen so that signals generated by individual sensors, from a sound source 54 of interest emitting a plurality of frequencies, are distinguishable in terms of phase and power, while allowing the sensors to be in generally the same sound field, i.e., to be comparably affected by the same sound sources. When phase information is used, distances between sensors 52 must also be small enough to prevent spatial aliasing in the received range of frequencies. Most preferably, sound source 54 is at a position within a middle distance range of approximately 3-100 cm from the center of array 50.

[0059] If sensors 52 are unidirectional, they are preferably oriented in substantially identical directions towards sound source 54. Alternatively, one of the sensors, preferably the sensor farthest from sound source 54, may be oriented in the opposite direction in order, inter alia, to improve discrimination between sound coming from the direction of the sound source and sound coming from an opposite direction.

[0060] Fig. 4 is a schematic block diagram illustrating the operation of processor 60, in accordance with a preferred embodiment of the present invention. Processor 60 may be implemented in an analog, a digital or a hybrid analog-digital form, by methods of electronic design and, as appropriate, software programming known in the art. Processor 60 comprises a master signal generator 62, a set of splitters 64, a gain controller 66, a gain reducer 68, a combiner 70, and a master signal splitter 72. Preferably, when processor 60 is implemented in a digital form, processor 60 comprises a plurality of analog-to-digital converters 74 and a digital-to-analog converter 76. Converters 74 digitize signals from sensors 52 and transfer the respective digitized signals to splitters 64 and master signal generator 62. Digital-to-analog converter 76 outputs the signal from combiner 70 as an analog signal. Alternatively, when processor 60 is implemented in an analog or in a hybrid analog-digital form, signals from sensors 52 are transferred directly to splitters 64 and master signal generator 62, and combiner 70 generates an analog output directly.

[0061] Each splitter 64 comprises a set of band-pass filters, which receive signals from a respective sensor 52, split the signals into frequency bands using the filters, and transfer the split signals to gain controller 66. In an analog or hybrid implementation, splitters 64 are implemented as a set of analog filters. Preferably, each filter covers approximately 1/3 of an octave, and the set is sufficient to cover the audible range of sound. In a digital implementation, splitters 64 are preferably implemented as a sequence of windowed FFT transforms with half-window overlap performed on every individual signal. Hanning windows, as are known in the art, are preferably used for this purpose. The sequences of complex coefficients resulting from the FFTs represent individual frequency band signals.

[0062] Master signal generator 62 receives full-band signals from each sensor 52 and generates one output master signal. Master generator 62 preferably produces the output master signal such that the sound source of interest is enhanced compared to the signals from individual sensors, utilizing any suitable beam-forming strategy known in the art. For example, *in Numerical Optimization of Non-adaptive Microphone Arrays, by* Alexander Goldin, Proc. IEEE Int. Conf. on Acoustics, Speech and Signal Processing (1997), pages 507-510, the author describes how an enhanced signal may be generated as a sum of filtered signals from individual sensors. The filters are fixed and chosen in an array design stage during a numerical optimization procedure. The purpose of the optimization is to provide good directional characteristics with minimal off-axis frequency coloration of the enhanced signal. Alternatively, generator 62 may just use the signal from one of sensors 52, preferably a sensor oriented towards the sound source of interest, without altering the signal.

[0063] The output signal from generator 62 is split by master signal splitter 72 into frequency bands to generate a master set of signals, which is output to gain controller 66. Preferably, master splitter 72 is constructed and functions in substantially the same manner as each of splitters 64.

[0064] Controller 66 operates by setting the gain of each particular frequency band according to a function of parameters of the master signal and the signals from individual sensors 52. A generalized gain function is given by the following equation:

$$G_i(t) = F\left(p_1^M(t),\ldots,p_1^M(t),p_1^1(t),\ldots,p_1^1(t),\ldots,p_1^m(t),\ldots,p_1^m(t)\right) \quad (1)$$

Here $G_i(t)$ is the gain for band i at an instant t, k is the number of bands into which the signal is split, m is the number of sensors in array 50, $p_j^M(t)$, $j=1,..,k$, are instantaneous parameters of the master signal, and $p_j^n(t)$, $j=1,..,k$ are instantaneous parameters of the signal from the n^{th} sensor. Typically, the parameters represent respective amplitudes, powers, or phases of the signals in the given frequency bands.

[0065] Gain controller 66 generates a gain for each of the k frequency bands depending on short-term power and phase information in the input signals. (A specific gain function is described in detail hereinbelow, with reference to Figs. 6, 7, and 8, whereby gain controller 66 assigns gains in the different bands produced by splitters 64 and splitter 72 according to whether the signal in a particular band comes from inside or outside a "sensitivity region," in front of array 50. In general, signals originating outside the sensitivity region are suppressed.) Each gain is directly applied to the respective master signal band in gain reducer 68, which multiplies each band by its respective gain. The separate output band signals are then combined to form one output signal in combiner 70. For an analog or hybrid implementation, combiner 70 is an analog mixer. For a digital implementation combiner 70 is preferably a simple digital mixer. If a FFT is used in splitters 64 and 72, then an inverse FFT is used in combiner 70. Alternatively, combiner 70 may have a more complex structure, as is known in the art.

[0066] Fig. 5 is a schematic block diagram showing the operation of a processor 80, in accordance with an alternative preferred embodiment of the present invention. Apart from the differences described hereinbelow, processor 80 operates in substantially the same manner as processor 60, whereby components having the same numbers in processors 60 and 80 are constructed and function in substantially the same manner. Signals from sensors 52 are transferred to corresponding splitters 64. The transfer is direct if processor 80 is constructed as an analog or a hybrid embodiment. The transfer is via A/D converters 74 if processor 80 is a digital embodiment. After splitting the respective signals into frequency bands, each splitter transfers the split signals to gain controller 66 and to master signal generator 62. Generator 62 combines the split signals, using band-by-band addition or other processes known in the art, to generate a master set of signals in the frequency bands corresponding to those of splitters 64. The master set of signals is then transferred directly to gain controller 66 and gain reducer 68, so that master splitter 72 of processor 60 is not required. The remainder of the operation of processor 80 is substantially as described above with reference to processor 60.

[0067] Fig. 6 is a schematic diagram showing the layout of a microphone array 90, applicable to a preferred embodiment of the present invention. The diagram illustrates a method for using instantaneous sound powers as the parameters of equation (1), to calculate a corresponding gain function. Array 90 comprises two omnidirectional sound sensors 92 lying on an axis 96 and separated by a distance $2D$. A sound source 94 is located a distance L from the center of array 90, and subtends an angle θ with axis 96. The distances L_1, L_2 between sound source 94 and the sensors are given by:

$$L_1 = \sqrt{(L\sin(\theta))^2 + (L\cos(\theta)-D)^2}$$

$$L_2 = \sqrt{(L\sin(\theta))^2 + (L\cos(\theta)+D)^2}$$

[0068] The angles θ_1 and θ_2 between axis 96 and the lines connecting sound source 94 to sensors 92 are given by:

$$\theta_1 = \arctan\left(\frac{L\sin(\theta)}{L\cos(\theta) - D}\right)$$
$$\theta_2 = \arctan\left(\frac{L\sin(\theta)}{L\cos(\theta) + D}\right) \qquad (2)$$

[0069] According to the inverse square law, as applied to a sound source radiating a substantially spherical wave, the instantaneous power of a signal is inversely proportional to the square of the distance to the source. Thus for sensors 92, the ratio *r* of the two powers P_2 and P_1, having respective amplitudes A_2 and A_1, at the sensors is given by:

$$r = \frac{P_2}{P_1} = \frac{A_2^2}{A_1^2} = \frac{L_1^2}{L_2^2} = \frac{(L\sin(\theta))^2 + (L\cos(\theta)-D)^2}{(L\sin(\theta))^2 + (L\cos(\theta)+D)^2} \qquad (3)$$

[0070] For a finite $L>D$:

r < 1 when -90° < θ < 90°, i.e., the sound source is in front of the array.
r = 1 when θ = ±90°, i.e., the sound source is transversal to the array.
r > 1 when 90° < θ < 270°, i.e., the sound source is behind the array.
r also approaches 1 for all angles θ when the distance *L* is large compared to D.

[0071] Thus, the inequality

$$r = \frac{P_2}{P_1} < r_c,\ r_c < 1 \qquad (4)$$

defines a region in space, herein termed a sensitivity region, wherein r_c is a "critical ratio."

[0072] Fig. 7 is a graphical (L, θ) plot illustrating sensitivity regions for array 90, according to a preferred embodiment of the present invention. The plots are calculated using the geometry for array 90 shown in Fig. 6, with *D* = 40 mm. Array 90 is positioned at the center of a 500 mm radius circle 102. An outer line 100 encloses a sensitivity region 104 defined by $r < r_c$ wherein $r_c = 0.7$. A middle line 110 encloses a sensitivity region 106 defined by r < 0.6, and an inner line 120 encloses a sensitivity region 108 defined by r < 0.4. (Lines 100, 110, and 120 are generated by solving equation (3) for L for the respective values of r_c.) For example, if the sound power ratio ($r = P_2/P_1$) for sensors 92 is found to be 0.65, then the source generating the sound is within region 104, and is external to regions 106 and 108. Thus, if it is found that $r<r_c$ for a particular frequency band of sound received by array 90, then the source of the sound is considered to be within the region, so that the gain G for the particular band should be theoretically set to unity. If $r>r_c$, then the sound source is assumed to be outside the region, so that the gain G should be theoretically set to zero.

[0073] Fig. 8 is a graph showing theoretical and actual gain functions when $r_c = 0.6$, according to a preferred embodiment of the present invention. A line 130 in the graph corresponds to a theoretical gain step function {(r,G)|G = 1, r<0.6; G = 0, r>0.6}, fitting the model described above with reference to Fig. 7. Preferably, however, the step function is modified to have a generally continuous transition region 134, as illustrated by a curve 132.

[0074] Referring back to Figs. 4 and 5, gain controller 66 calculates the ratio r of instantaneous powers in a particular frequency band. Preferably, controller 66 utilizes a gain function of the form shown by line 132, Fig. 8, to set the gain of each band. The gains are applied to each band of the master signal in gain reducer 68, and the signals produced are combined in combiner 70, as described above, to produce the output signal.

[0075] In an alternative preferred embodiment of the present invention, each sensor 92 (Fig. 6) is a unidirectional pressure sensor having a gain function of the form G = T(θ), wherein θ is the angle of incidence of the signal at the sensor. Using unidirectional sensors makes decision-making easier, by initially attenuating interference that otherwise may adversely affect the ratio of energies when both the interference and the signal are present. This may significantly improve the quality of the output signal. Equation (3) then becomes:

$$r = \frac{P_2}{P_1} = \frac{A_2^2}{A_1^2} = \frac{T^2(\theta_1)L_1^2}{T^2(\theta_2)L_2^2} = \frac{T^2(\theta_1)}{T^2(\theta_2)} \bullet \frac{(L\sin(\theta))^2 + (L\cos(\theta)-D)^2}{(L\sin(\theta))^2 + (L\cos(\theta)+D)^2} \qquad (5)$$

The angles θ_1, θ_2 are computed according to equation (2). As will be clear to those skilled in the art, equation (5) can be used to generate a gain function corresponding to equation 1.

[0076] While the example described above with reference to Figs. 6, 7, and 8 uses two sensors, the method embodied in this example can be generalized in a straightforward manner to enable similar calculations to be performed for three or more sensors in array 90, and thus to produce respective gain functions corresponding to equation 1. Gain functions generated for three or more sensors are generally more stable when random fluctuations of the sound field occur in a closed environment. Furthermore, master signals produced by three or more sensors are generally enhanced compared to master signals produced by two sensors. For example, an average or maximal ratio between signal powers from adjacent sensors may be used to compute the gain function. Increasing the number of sensors is known to improve the directivity of the fixed array used to generate the master signal.

[0077] The example described with reference to Figs. 6, 7, and 8 uses instantaneous sound pressures as parameters for determining the gain function of equation (1). Alternatively or additionally, instantaneous phase differences between band signals may also be utilized to determine the gain function. In a preferred embodiment of this type, the maximum allowable value of a phase difference from a source within a particular angular sector for a specific band, corresponding to the critical ratio of equation (4), is computed. A gain function similar to that described with reference to Fig. 8 is generated using phase differences as parameters, and is applied by gain controller 66 so as to generate the output signal. Generally, the gain in a given band is reduced if the actual phase difference is greater than the maximum allowable value.

[0078] Furthermore, the present invention is not limited to the specific gain functions described hereinabove, and it will be appreciated that other gain functions, based on sound power, phase and/or other parameters, may also be used in microphone systems for the purpose of discriminating between sound sources by their distance range from the microphones.

[0079] It will be further appreciated that the preferred embodiments described above are cited by way of example, and the full scope of the invention is limited only by the claims.

Claims

1. A method for enhancing discrimination of sound received from a sound source (54) relative to acoustic interference, comprising:

   providing a plurality of sound sensors (52) in predetermined positions;
   receiving respective signals from the plurality of sound sensors responsive to the interference and to the sound source;
   determining respective characteristics of the plurality of signals in each of a plurality of spectral bands;
   analyzing the determined characteristics to compute a spectral gain function (132) which discriminately enhances a portion of the signals that is associated with the sound source;
   processing the signals from one or more of the plurality of sensors to generate a combined master signal; and
   applying the spectral gain function to the master signal so as to generate an output signal in which the portion of the signals associated with the sound source is enhanced relative to that due to the acoustic interference.

2. A method according to claim 1, wherein applying the spectral gain function comprises splitting the master signal into a plurality of spectral bands corresponding to the plurality of bands with respect to which the characteristics are determined, and applying a gain factor to each of the bands.

3. A method according to claim 2 or 3, wherein analyzing the determined characteristics comprises determining a gain function responsive to a power difference of the signals received from the sound sensors.

4. A method according to any of the above claims, wherein analyzing the determined characteristics comprises determining a gain function responsive to a phase difference of the signals received from the sound sensors.

5. A method according to any of the above claims, wherein processing the one or more signals to generate the master signal comprises summing respective spectral components of the one or more signals in at least one frequency

band.

6. A method according to any of the above claims, wherein processing the one or more signals to generate the master signal comprises combining the signals responsive to relative phases thereof so as to enhance a contribution to the master signal of sound coming from a preferred direction.

7. A method according to any of the above claims, wherein receiving the respective signals compresses using a Fast Fourier Transform (FFT), and wherein applying the spectral gain function comprises using an inverse FFT.

8. A method according to any of the above claims, wherein analyzing the determined characteristics comprises selecting a sensitivity region within which the sound source is detected.

9. A method according to claim 8, wherein the sensitivity region comprises distances in a range of 3-100 cm from the plurality of sound sensors.

10. A method according to any of the above claims, wherein the plurality of sensors comprises at least one omnidirectional sensor.

11. A method according to any of the above claims, wherein the plurality of sensors comprises at least one unidirectional sensor.

12. A method according to claim 11, wherein computing the gain function comprises computing the function responsive to a unidirectional sensor gain function.

13. A method for enhancing discrimination of sound received from a source (54) in a given location relative to acoustic interference, comprising:

providing an array of sound sensors (52) in a predetermined position;
receiving respective signals from the array of sound sensors responsive to the interference and to the sound from the source;
analyzing the signals to identify one or more characteristics of sound received from within a selected range of distances that includes the distance of the location of the source from the position of the array;
determining a gain function (132) responsive to the identified characteristics; and
applying the gain function to the received signals so as to generate an output signal in which a portion of the signals corresponding to sound received from within the selected range of distances is enhanced relative to sound from outside the range.

14. A method according to claim 13, wherein determining the gain function comprises determining a gain function responsive to a power difference of the signals received from the sound sensors.

15. A method according to claim 13 or claim 14, wherein determining the gain function comprises determining a gain function responsive to a phase difference of the signals received from the sound sensors.

16. A method according to any of claims 13-15, wherein analyzing the signals comprises determining respective characteristics of the signals in each of a plurality of spectral bands.

17. A method according to claim 16, wherein determining the gain function comprises determining a gain function using at least one of the spectral bands and applying the function to the other bands.

18. A method according to any of claims 13-17, wherein analyzing the signals comprises using a Fast Fourier Transform (FFT), and wherein applying the gain function comprises using an inverse FFT to generate the output signal.

19. A method according to any of claims 13-18, wherein the array of sensors comprises at least one omnidirectional sensor.

20. A method according to any of claims 13-19, wherein the array of sensors comprises at least one unidirectional sensor.

21. A method according to claim 20, wherein determining the gain function comprises computing the function responsive to a unidirectional sensor gain function.

22. Apparatus for enhancing discrimination of sound received from a sound source (54) relative to acoustic interference, comprising:

a plurality of sound sensors (52) which generate a respective plurality of signals responsive to the interference and to the sound source;
a plurality of splitters (64), which divide the respective plurality of signals into a plurality of spectral bands;
a master signal generator (62), which generates a master signal responsive to at least one of the plurality of signals;
a gain controller (66), which computes a spectral gain function which discriminately enhances a portion of the signals that is associated with the sound source responsive to the signals; and
a signal combiner (70), which applies the spectral gain function to the master signal so as to generate an output signal in which the portion of the signals associated with the sound source is enhanced relative to that due to the acoustic interference.

23. Apparatus according to claim 22, wherein the master signal generator comprises a splitter (72) which splits the master signal into a plurality of spectral bands corresponding to the plurality of bands into which the plurality of splitters divide the signals.

24. Apparatus according to claim 22 or claim 23, wherein the gain controller computes the gain function responsive to a power difference of the signals received from the sound sensors.

25. Apparatus according to any of claims 22-24, wherein the gain controller computes the gain function responsive to a phase difference of the signals received from the sound sensors.

26. Apparatus according to any of claims 22-25, wherein the plurality of sound sensors comprises at least one omnidirectional sensor.

27. Apparatus according to any of claims 22-26, wherein the plurality of sound sensors comprises at least one unidirectional sensor.

28. Apparatus according to claim 27, wherein the gain controller computes the spectral gain function responsive to a unidirectional sensor gain function.

29. Apparatus for enhancing discrimination of sound received from a source (54) in a given location relative to acoustic interference, comprising:

an array of sound sensors (52) in a predetermined position, which sensors generate a respective plurality of signals responsive to the interference and to the sound source;
a gain controller (66), which analyzes the signals to identify one or more characteristics of sound received from within a selected range of distances that includes the distance of the location of the source from the position of the array, and which determines a gain function responsive to the identified characteristics; and
a signal combiner (70), which applies the gain function to the received signals so as to generate an output signal in which a portion of the signals corresponding to sound received from within the selected range of distances is enhanced relative to sound from outside the range.

30. Apparatus according to claim 29, and comprising a plurality of splitters (64), which respectively split the signals received from the sensors into a plurality of spectral bands.

31. Apparatus according to claim 29 or claim 30, and comprising a master signal generator (62), which generates a master signal responsive to at least one of the plurality of signals, and to which master signal the signal combiner applies the gain function.

32. Apparatus according to any of claims 29-31, wherein the array of sound sensors comprises at least one omnidirectional sensor.

33. Apparatus according to claim any of claims 29-32, wherein the array of sound sensors comprises at least one unidirectional sensor.

34. Apparatus according to claim 33, wherein the gain controller computes the gain function responsive to a unidirectional sensor gain function

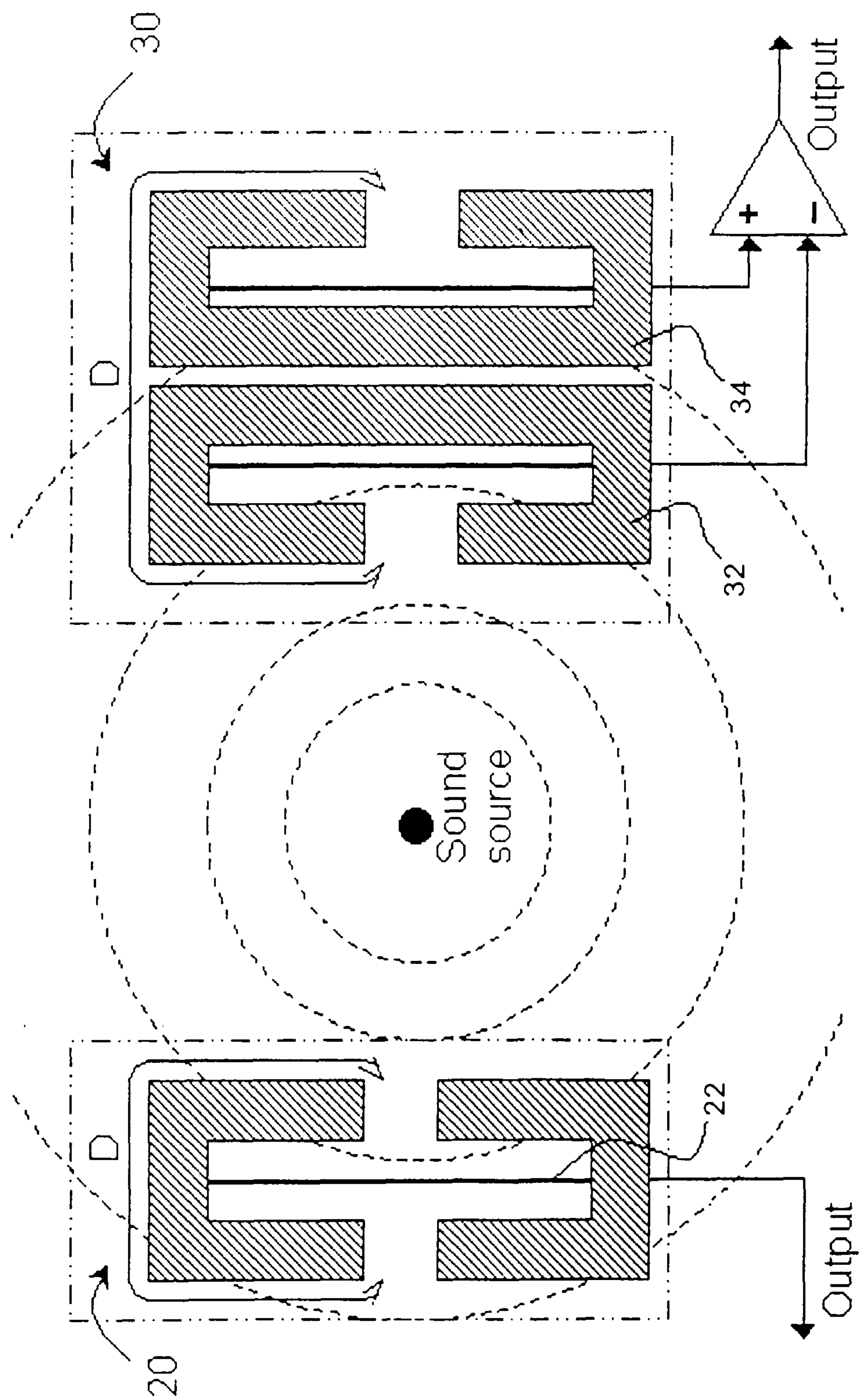


Fig. 1 Prior Art

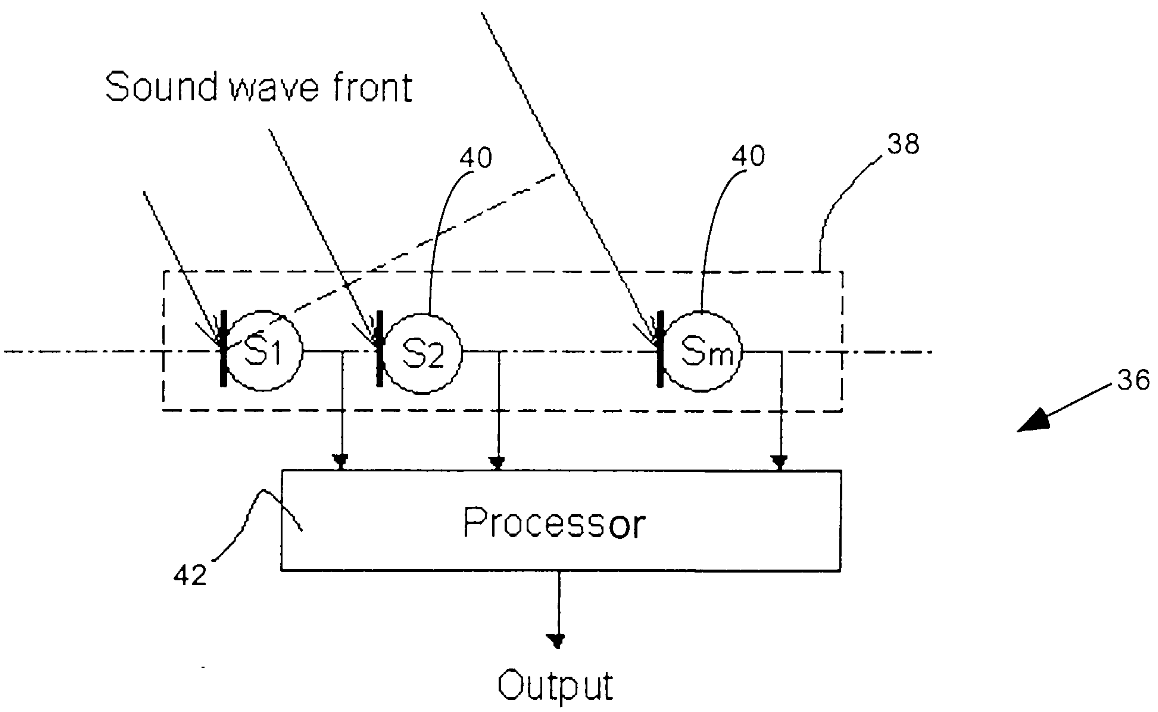


Fig. 2 Prior Art

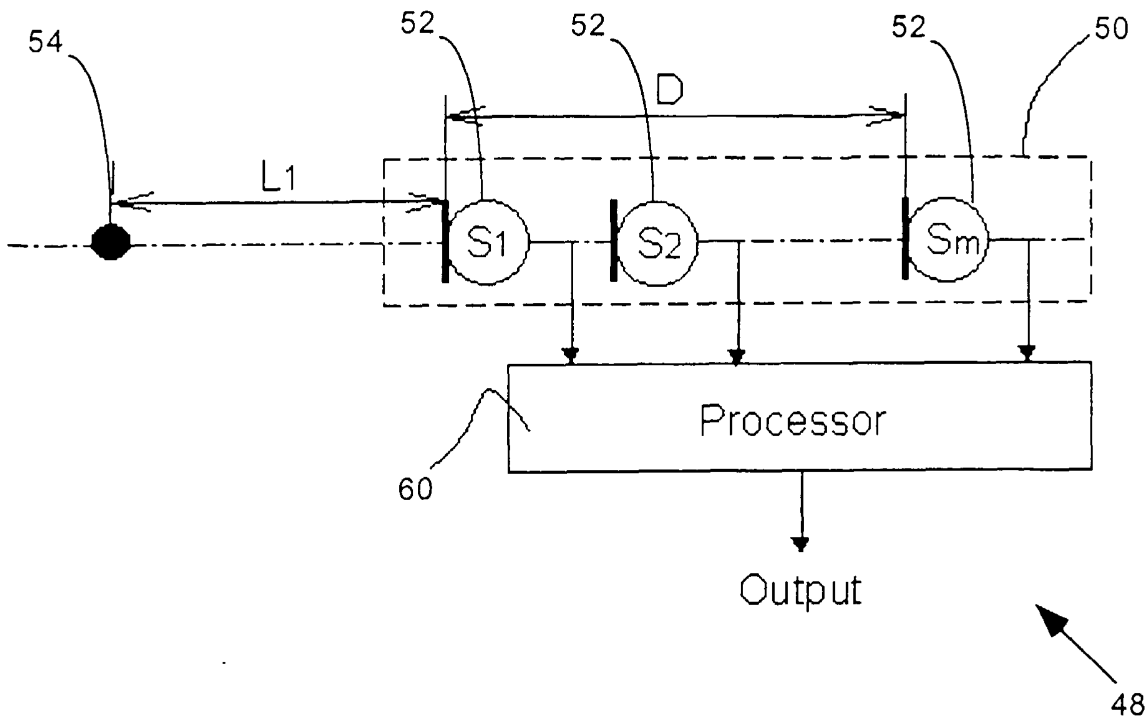
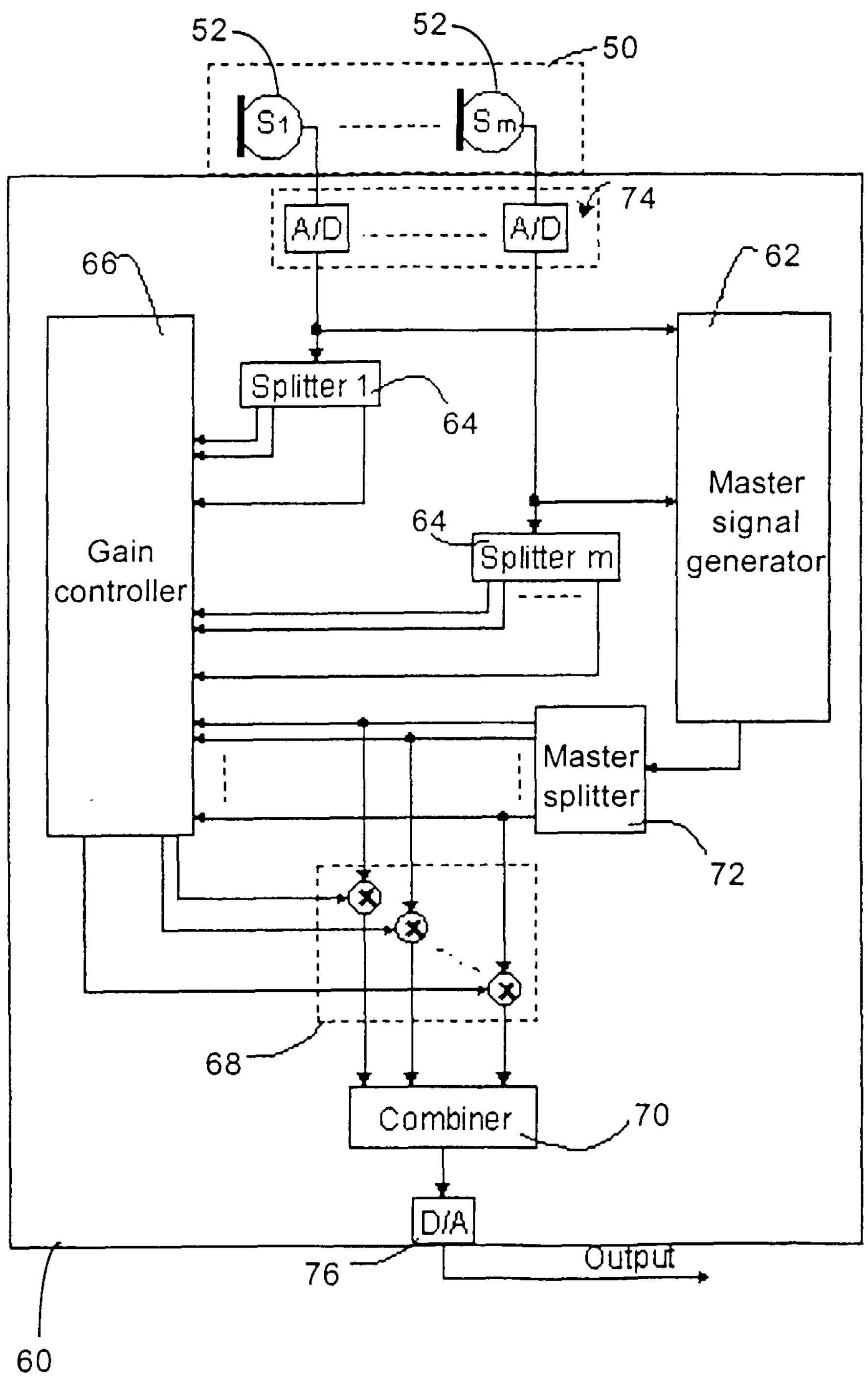


Fig. 3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50
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Sm
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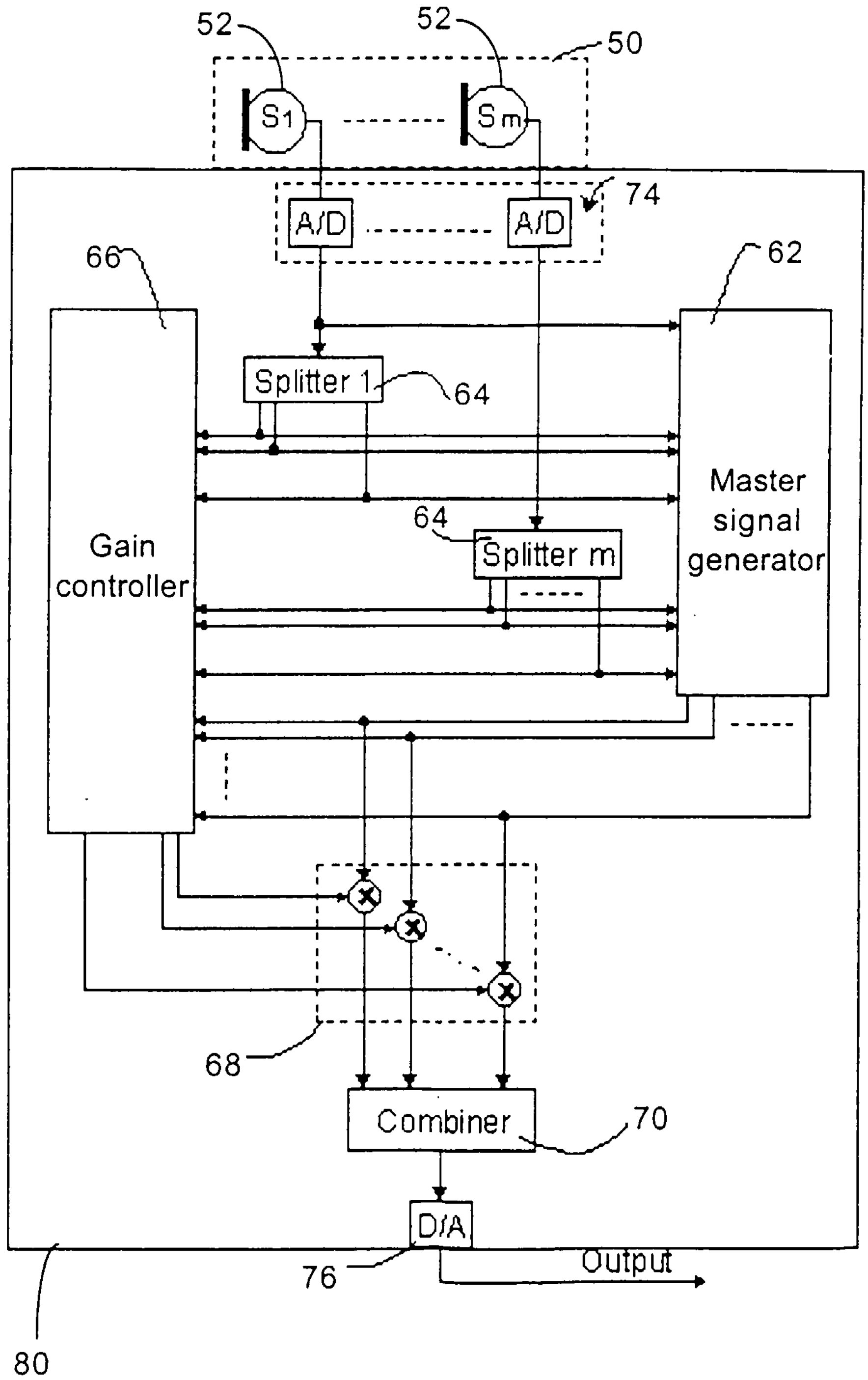
A/D
A/D
66
62
Splitter 1
64
Gain
controller
Master
signal
generator
64
Splitter m
Master
splitter
72
68
Combiner
70
D/A
76
Output
60

Fig. 4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A/D
66
62
Splitter 1
64
Master
signal
generator
64
Gain
controller
Splitter m
68
Combiner
70
D/A
76
Output
80

Fig. 5

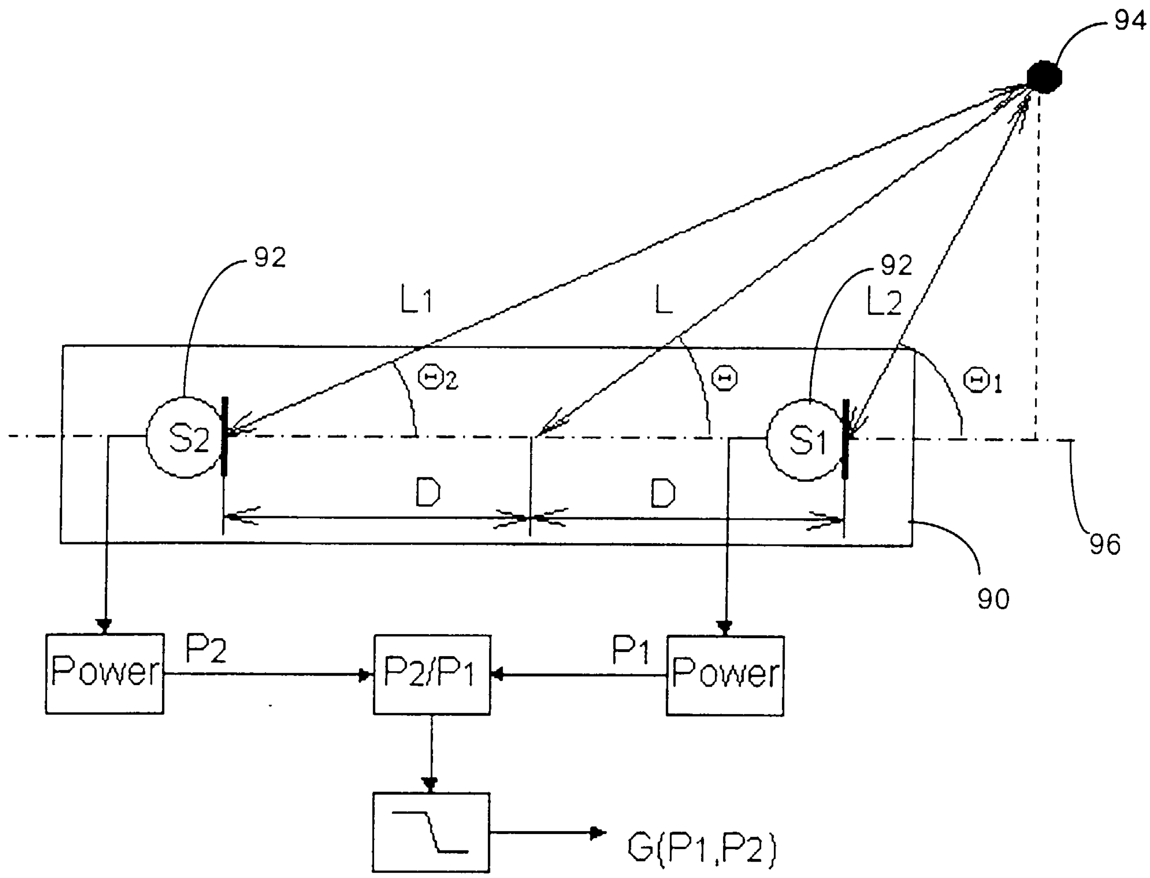
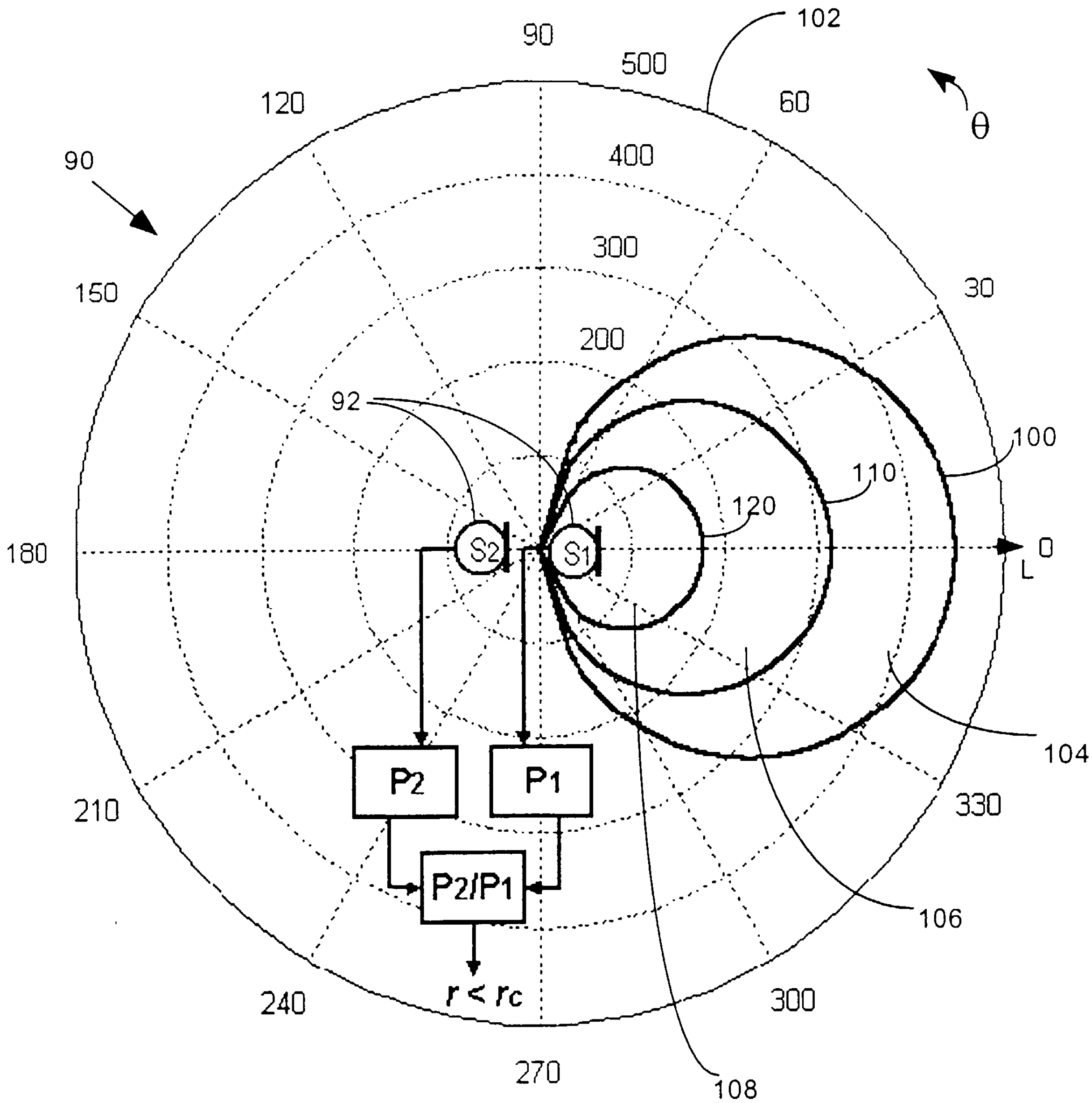


Fig. 6



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θ
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300
150
30
200
92
100
110
120
180
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S1
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L
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P2
P1
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330
P2/P1
106
240
r < rc
300
270
108

Fig. 7

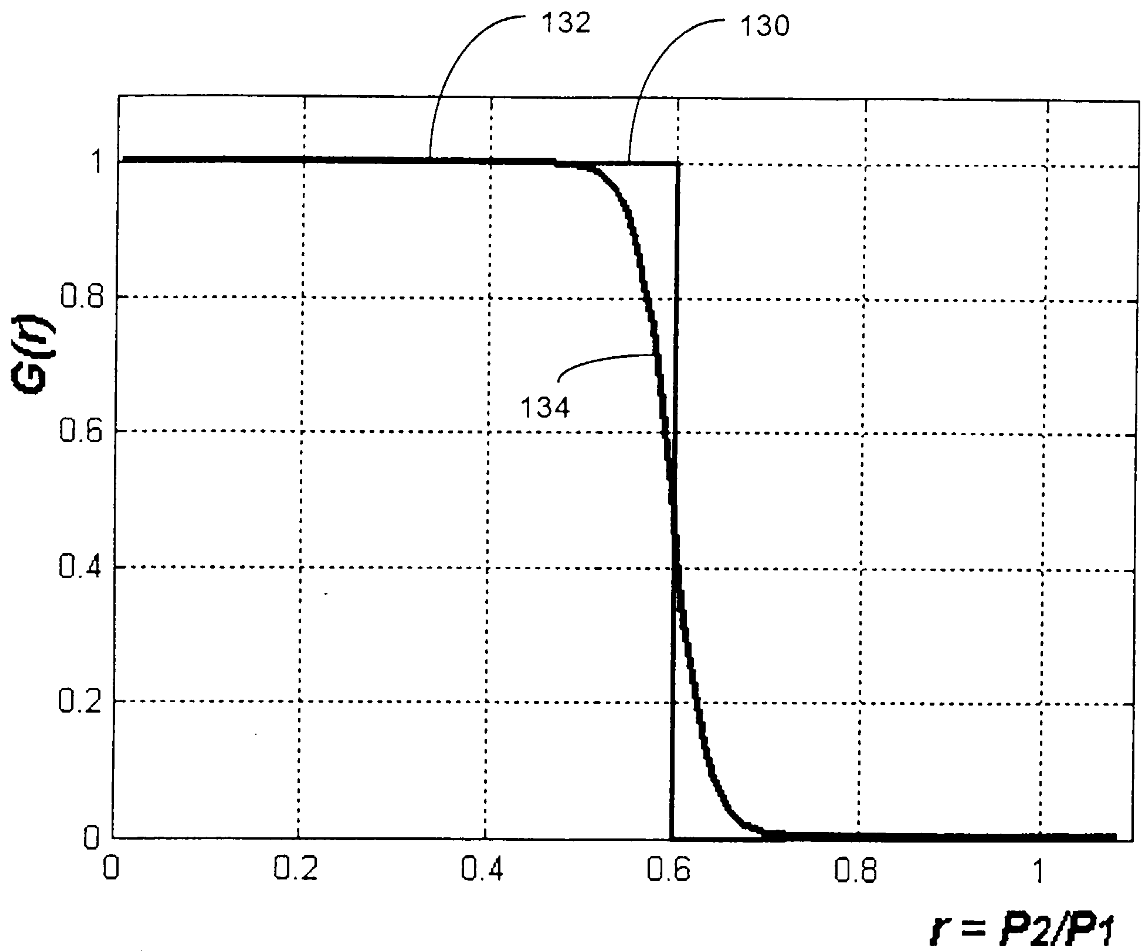

Fig. 8