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Kryter

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[54] **METHOD OF AND APPARATUS FOR AIDING HEARING AND THE LIKE**

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[51] **Int. Cl.** **H04r 25/00**

[58] **Field of Search** ..... 179/107 FD, 107 R, 107 S, 179/1 D, 1 FS

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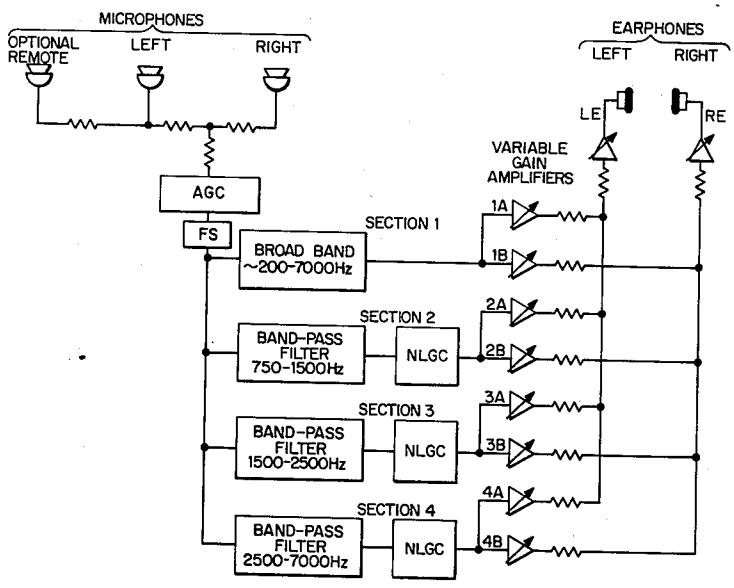
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[57] **ABSTRACT**

This disclosure deals with electronically aiding sensori-neural deafness with frequency-segmented, dynamic range-compressed speech signal processing, wherein noise vs. speech signal discrimination is employed with an optional semi-remote microphone input, and with an optional electronic frequency-shift processing of the signal to prevent or reduce oscillation due to acoustic airborne and/or vibrational feedback between the earphone(s) and the microphone(s).

12 Claims, 2 Drawing Figures



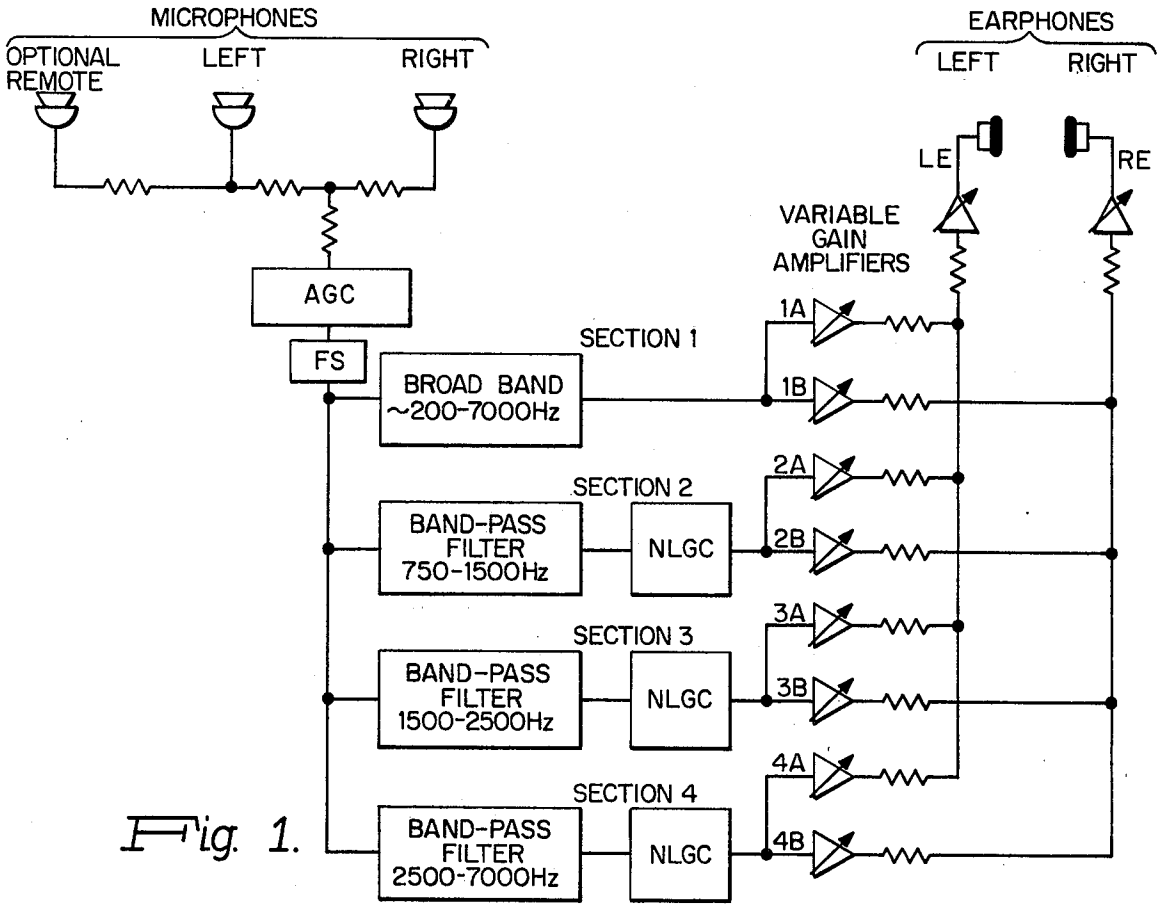

MICROPHONES
OPTIONAL REMOTE
LEFT
RIGHT
EARPHONES
LEFT
RIGHT
LE
RE
VARIABLE GAIN AMPLIFIERS
AGC
FS
SECTION 1
BROAD BAND ~200-7000Hz
1A
1B
SECTION 2
BAND-PASS FILTER 750-1500Hz
NLGC
2A
2B
SECTION 3
BAND-PASS FILTER 1500-2500Hz
NLGC
3A
3B
SECTION 4
BAND-PASS FILTER 2500-7000Hz
NLGC
4A
4B

Fig. 1.

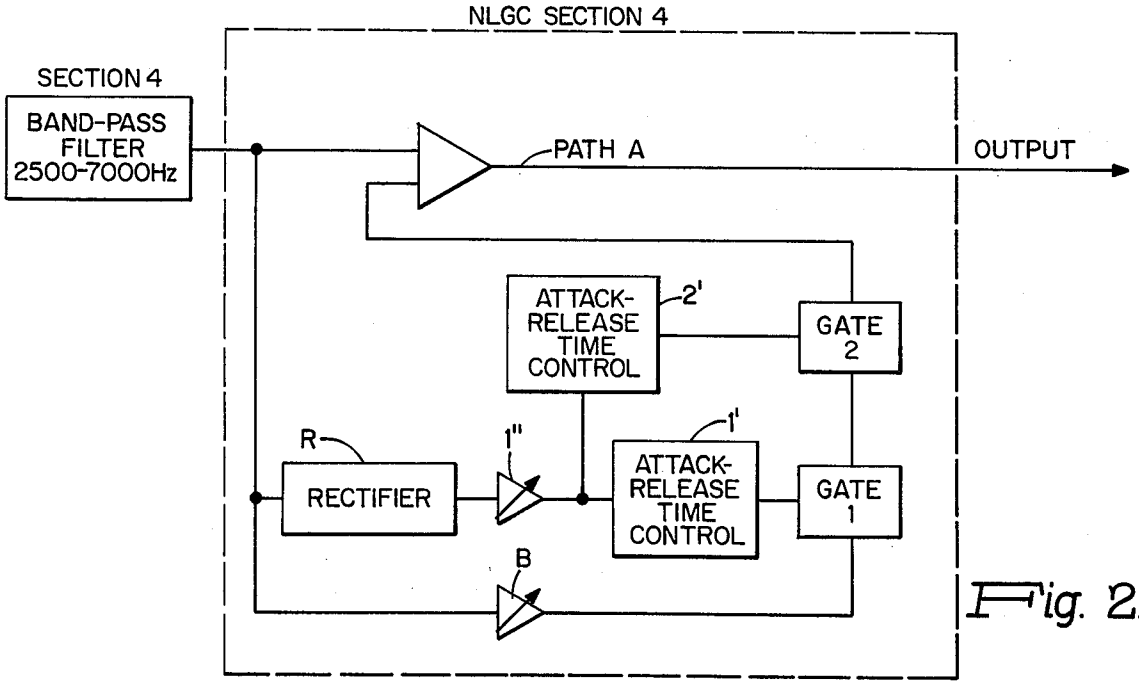

NLGC SECTION 4
SECTION 4
BAND-PASS FILTER 2500-7000Hz
PATH A
OUTPUT
ATTACK-RELEASE TIME CONTROL
2'
GATE 2
R
RECTIFIER
1"
ATTACK-RELEASE TIME CONTROL
1'
GATE 1
B

Fig. 2.

METHOD OF AND APPARATUS FOR AIDING HEARING AND THE LIKE

The present invention relates to methods of and apparatus for electronically aiding hearing or similar applications, being more particularly directed to improving noise vs. speech signal discrimination.

The most prevalent type of deafness is so-called sensori-neural hearing loss, wherein the inner ear loses some ability to perceive the weaker intensity portions of the speech signal and also loses some ability to make normal discriminations among some frequency components even though of sufficient intensity to be audible to the person with sensori-neutral hearing loss. Usually these losses in hearing ability are greater for the higher sound frequencies, say, 2000 Hertz) than for the lower (below, say 2000 Hertz). The sensori-neural deafened ear, moreover, causes the perception of sounds that are very intense as excessively loud. Distortions not formed in the normal inner ear, which contains the sensori-neural receptors, moreover, apparently occur in the sensori-neural deafened ear and result in less discrimination than normal among the various speech sounds.

There are many electronic hearing aids which provide means for increasing the intensity of the speech signal reaching the inner ear so that the weakened sounds are audible to the deafened ear. These hearing aids, however, while of help to persons suffering so-called conductive type deafness, are not very helpful to sensori-neural deafness because of the aforementioned loss in discrimination ability, and because of the inner-ear distortions and excessive loudnesses that occur when sound amplification is applied to the strong as well as weak components of the varying intensity speech signal in order that the weaker sounds be made audible to the sensori-neural ear. For example, a word such as "show" contains the consonant "sh", which is much weaker than the vowel sound "ow". A hearing aid that sufficiently amplifies all the sounds uniformly or linearly so that the weaker "sh" component, or "phoneme", as it is called, is audible to the sensori-neural ear, may also make the "ow" portion of the word extraordinarily loud and cause distortion in the inner ear, thereby tending to lessen understanding of the speech signal. It is also important to note that these weaker phonemes tend to have durations ranging from about 0.01 to less than 0.5 second. It has been discovered, in accordance with the present invention, that effective use can be made of the relative difference in amplitude of segments of the speech signal and the relatively short duration of the speech segments of phonemes, particularly the less intense phonemes, to produce the improved results herein described.

In attempts to overcome the deficiency of linear-gain hearing aids, automatic non-linear or compression gain control systems have sometimes been used wherein the intensity of the speech signal is averaged for a brief period of time and this information is used automatically to adjust the gain of the amplifier. If the level is too low, the gain of the amplifier is increased by an amount proportional to the degree the average input voltage (over some specified period of time) falls below a specified level. This process is called dynamic range compression; but it is difficult satisfactorily to achieve with speech signals because the signal level changes so quickly from one speech sound to another. Changing the gain without an adequate determination of the average envelope will cause distortion of the signal waveform and thereby degrade its understandability. In brief, an automatic gain control system that more or less continuously (or too frequently) modifies the degree of gain will tend to introduce distortion and as a result will not always make the speech signal more understandable, as described by E. Trinder, An Attempt to Correct Speech Discrimination Loss in Cochlear Deafness by Graded Instantaneous Compression, *Sound*, Vol. 5, pp. 62–67, (1972). Conversely, maintaining a given gain for too long a period of time will also degrade the understandability of the speech signal because the gain setting will be inappropriate over significant segments of the speech phonemes wherein the level changes are very rapid.

Another shortcoming of automatic compression gain control systems is that during periods of time when there is a pause in the input speech signal, the gain control is progressively increased to a maximum amount and thereby tends to make objectionable to the hearing aid user, the normally low level, or residual, noise present at the input of inherent in the electronics of the hearing aid. It is noted that in the present hearing aid invention, as will be described later, an automatic non-linear-linear gain control (to be labelled NLGC) device is utilized that has the ability to discriminate to a degree between speech signals and background noise and adjust the system gain appropriately on the basis of this information; i.e., prevent excessive amplification to the weak noise segments.

It might be noted that some reduction in the distortions that occur with automatic compression gain can be reduced to some extent by the application of independent automatic compression gain controls to different portions of the speech spectrum; the amount, if any, for each portion being adjusted to meet the degree and kind of hearing loss experienced by a given ear with a sensori-neural hearing loss. Such automatic compression gain of frequency segment speech signals has been described, for example, by E. Villchur, Signal Processing to Improve Speech Intelligibility in Perspective Deafness, *J. Acoust. Soc. Am.* 53, 1647–1657, (1973). While this technique does provide improvement in understanding of speech by persons with sensori-neural deafness, it does not provide for the discrimination between weak speech segments and weak noise segments providing increased gain for the speech segments but not the noise segments, as does the present invention.

It is well known that persons wearing hearing aids with microphones, either non-directional or so-called directional located on or near the head of the listener, have difficulty in understandidng speech when in a conference or other situation where several speech or other competing auditory signals reach the listener at about the same time. This difficulty can be partly overcome by orienting the listener's microphones, especially if they are of the directional type, as described, for example, in U.S. Pat. No. 3,770,911, so that they pick up the desired signal to a greater extent that the undesired signals because of acoustical reasons. An additional advantage, however, can be provided if the listener were to place a microphone nearer the source of the desired signal which would increase the intensity of this signal at the microphone pick-up relative to that of the other signals that are present. Under many social circumstances it would be appropriate to accomplish this without obvious and awkward movements on the

part of the listener using a hearing aid with such a movable microphone.

A common problem of hearing aids that are designed to provide large amounts of signal gain for persons with unusually large amounts of hearing loss is that some of the output of the earphones of the hearing aid "leaks" or feeds back either by air or by mechanical paths, to the microphone of the hearing aid. This feedback causes a cyclic reamplification or oscillation that leads to complete overloading of the hearing aid causing it to "squeal" and be obnoxious and useless to the user. A procedure for reducing a related type of oscillation, but in the different application and requirements of public-address systems operated in a reverberant room, has been described by M. R. Schroeder, "Improvement of Acoustic-Feedback Stability by Frequency Shifting," *J. Acoust. Soc.*, 36, 1718–1724, (1964).

In this procedure, the airborne signal picked up by the microphone is shifted, by well-known modulation techniques, either upwards or downwards by about 5 to 10 Hz before it is presented to the acoustic output transducers or loudspeakers of the public address system. This shift in frequency is not sufficient significantly to interfere with the audible quality of the signal, particularly if the signal is speech, coming from the loudspeakers but does allow the output signal to reach levels about 10 dB higher without causing feedback oscillation than is possible without the application of the frequency shift processing. This frequency shifting process, properly critically adapted, has not heretofore been utilized for the prevention or reduction of either the mechanical linkage or the acoustic airborne feedback that may be present in such hearing aids. Indeed, it is to be noted that in earlevel hearing aids wherein the microphone and earphone are mounted in the same case or are mechanically linked through tubes or wires, the oscillation present in high-gain hearing aids is more often caused by the mechanical vibrations than the airborne. It is readily appreciated, however, that shifting the frequency coming from the earphone will tend, to a significant degree, to prevent the vibrations in the mechanical connection between the earphone and microphone from progressively enlarging, that occurs when the gain of the amplifier of the hearing aid is cyclically reapplied to the same input/output frequency. In brief, the input signal cannot be added to itself following amplification by the hearing aid and feedback, as normally can cause oscillation, because the signal is changed in frequency each time it passes through the hearing aid system and will therefore have a waveform, of feedback, that is not consistently in phase with the input waveform as is required, within limits, to cause oscillation of the system.

An object of the present invention, accordingly, is to provide a new and improved method of and apparatus for electronic hearing aiding that shall not be so subject to the above-described limitations and disadvantages of prior techniques, but that, to the contrary, significantly increases noise vs. speech signal discrimination, particularly useful for sensori-neural deafness problems and the like.

A further object is to provide a novel hearing aid or similar improved speech or related apparatus of more general character; other and further objects being later discussed and more specifically delineated in the appended claims.

In summary, the present invention provides real-time operation with special automatic gain control signal processing for both the overall signal and also for different parts of the speech spectrum in ways that can be adjusted to best suit the needs of individual sensori-neural deafened ears that suffer different degrees and patterns, as a function of frequency, of hearing deficiencies. The aid of the invention provides means of inserting one or more fixed increases in linear gain to segments of the speech signal that fall below given levels relative to the gain provided to segments that fall above given levels. The amount of increased gains and the given levels below which they are to be inserted may be set separately for each of the different parts filtered from the speech spectrum. Further, the invention will automatically discriminate between segments of the signal that constitute speech sounds and those segments consisting of background noise and will apply extra gain to the speech semgents, but not to the noise segments. The invention also provides for so-called bi-ear listening where the treatment of the signal for each of the ears of the listener can be somewhat different, and further provides for pick-up, if desired, by two microphones of a stereo signal, in order to utilize the information found in so-called phase differences between speech and other signals as present at two microphones; one placed at the position or pick-up region of each ear. Further, the hearing aid of the invention provides for an optional "remote" microphone that can be used for pick-up of signals at points at a farther-than-normal distance from the user, i.e. closer to the sound source. Further, the aid of the invention provides for an optional electronic frequency-shift of the signal picked up at the microphone so that the signal output at the earphones is at a somewhat different frequency (about 10 Hz) than the signal picked up by the microphone either by airborne or mechanical agitations.

The invention will now be described with reference to the accompanying drawing,

FIG. 1 of which is a block diagram of a preferred apparatus embodying the invention; and

FIG. 2 is a similar diagram of a suitable NLGC (non-linear-linear gun control) apparatus for use in the system of FIG. 1.

In addition to the optional remote microphone, so-labelled, two microphones (left and right) are indicated in FIG. 1 as the typical pick-up sources of the signal input, although the system could operate with even but one microphone. While the microphones may be non-directional, they are preferrably of the directional type, such as those described in said U.S. Pat. No. 3,770,911. One microphone will normally be located near or on the right ear, and one on or near the left ear; and their two inputs are inter-connected at the input of an automatic gain control circuit, labelled AGC No. 1. The optional "remote" microphone may be worn strapped to the user's wrist so that it appears as a wrist watch or bracelet and can be placed closer to a desired signal source by movement of his hand and arm, or it may be incapsulated in a pen or pencil type case, not shown, that can be laid on a conference table with a retractable cord extending to the hearing aid amplifier. The amplifiers, batteries and associated electronics of the apparatus, moreover, may be enclosed in a case worn in a clothes pocket of the user or attached to his or her body or clothing in any convenient manner.

The major purpose of the AGC circuit is to adjust the over-all speech signal to an intensity level for the filtering and additional automatic gain control processes to follow, such that the subsequent system will not overload, but yet will be at a level adequate to give proper signal transmission. Generally, at a distance of about three feet from a talker, the weaker speech sounds in conversational speech are at a level of about 20 dB re 0.0002 microbar, and the more intense speech sounds in a conversational speech signal are of the order of 60 dB. A dynamic range of about 40 dB is accordingly present in a speech signal uttered at a constant and conversational level of effort. When the listener is closer to the talker, furthermore, or when the talker uses a higher than normal effort of speaking, the level of the speech signals may go up to 100 decibels or so.

The AGC circuit is adjusted to provide a decrease in gain when the signal envelope is above a specified level (typically 60 dB) for approximately 0.001 seconds. Conversely, whenever the envelope level falls below a specified level (typically 60 dB) for approximately 0.02 seconds, the gain of the system automatically assumes its normal state of gain and treats signals between about 20 to 60 dB input in a substantially linear fashion. The decreases in gain effected by AGC are proportional to the degree to which the speech envelope (averaged over about 0.001 second) exceeds the level equivalent at that point in the system to a speech level of about 50 dB at the input to the microphone(s). Thus, AGC adjusts the average gain so that speech at an intensity greater than about 50 dB at the microphone(s) will generally be placed within the optimum operation region of the filters and associated electronic components that follow, as hereinafter described.

The signal from AGC may be fed to the frequency shifter section, if used, shown at FS, prior to being fed to Sections **1, 2, 3,** and **4** of the hearing aid as shown in FIG. **1.** Such a frequency-shifter FS, by means of standard RF modulation and demodulation techniques, as of the type disclosed in said Schroeder article, for example, shifts the frequency spectrum of the signal picked up at the microphone by about 10 Hz. Accordingly, the frequency spectrum coming from the earphones of the hearing aid is shifted from its location on the frequency spectrum, from the location it occupied when picked up by the microphone, increasing by about 10 dB the tolerable level of the level of output from the earphones that can be reached before acoustic feedback between the earphones and microphones causes oscillation in the hearing aid amplifiers. It is to be further noted that this frequency shifting process will also tend significantly to reduce the vibrations that may be set up in the mechanical linkage between the earphone(s) and microphone(s) of the hearing aid, said vibrations, if sufficiently strong, being a source of causing oscillation and overload in th hearing aid.

The signal from AGC or the optional frequency shifter is fed to and processed by Sections **1, 2, 3,** and **4** of the hearing aid, as shown in FIG. **1.** Section **1** passes a broad band of frequencies and each of sections **2, 3** and **4** contains a narrow band filter of different adjacent bands, as later explained. Section **1** transmits the broadband signal over the range of about 200 Hz to 7000 Hz to the listener, with adjustment of its level made possible by means of variable gain amplifiers **1A, 1B,** LE, and RE. To this broad-band signal are added the outputs of Sections **2, 3,** and **4,** which have broadly similar functions but are individually adjustable in several regards. The purpose of these sections is to separate or filter the speech or other acoustic signals into relatively narrow bands of frequencies so that the respective bands can be amplitude-processed and gain-adjusted in ways that will enhance the understandability of speech and other acoustic signals for persons with sensori-neural deafness. As indicated above, in certain regions, usually the higher frequency regions, the ear with sensori-neural deafness will usually have a usable dynamic range of but 10 dB or so between levels that are inaudible and levels that overload the ear, as compared with a usable dynamic range of more than 60 dB for the normal ear. At other frequency regions, the dynamic range may be greater or less, depending on the particular pattern of damage to the sensori-neural receptors in the inner ear. The purpose of the Sections **2,** 3, and **4** is to provide the means of processing the different frequency bands of speech to the degree and in the way best suited for the hearing characteristics of a given ear suffering sensori-neural deafness, and to add these specially processed frequency bands to the normal, broadband signal being transmitted by Section **1,** as shown in FIG. **1.** It is to be understood that for some sensori-neural deaf ears, fewer or more than four such sections of signal processors will be required, or that the bandwidths of one or more of the sections indicated may be changed, and that the non-linear gain control processing to be later described may be inactivated in given sections.

The description to follow of the functioning of Section **4** of FIG. **1,** for example, will suffice to explain also the operation of Sections **2** and **3,** except that the frequency-bands, the amplitude levels to which the gain is specially adjusted, and the following gain settings may be at different values for each section.

The band-pass filter of Section **4** separates the energy in the frequency band 2500 to 7000 Hz from the total spectrum of the signal. The output of this band-pass filter is then passed through a nonlinear-linear compresser gain control (NLGC). The amount of signal compression is set to be suited to the loss in a given ear in dynamic range of hearing ability for sounds in the frequency band 2500 to 7000 Hz. The NLGC operates such that when the signal is, for about 0.005 seconds, below a given level, an additional amount of signal energy is applied to the signal energy in the frequency band 2500 to 7000 Hz.

The output of this NLGC circuit is then further amplified in separate split paths containing amplifiers **4A** and **4B,** if necessary, to meet possible differences in sensitivity between the left and right ears of the listener.

Sections **2** and **3** are also individually separately adjusted to provide the degree, if any, of signal dynamic range compression best suited for optimizing the reception and understandability of signals, especially speech, as determined by the contributions of the several respective different frequency bands 750–1500 Hz and 1500–2500 Hz. The outputs of these three sections are then split into pairs and combined through resistor networks with the broad-band signal from Section **1** for presentation to the listener, with all the right and left ear paths of the NLGC outputs being connected together, respectively.

It is to be understood that the specific number of filter sections and the cut-off frequencies given in FIG. **1**

are illustrative only, and that greater or fewer sections and different cut-off frequencies may be used in various specific applications of this invention. Further, the use of separate output amplifiers for each of the two ears is often not required, because both ears of a person suffering sensori-neural deafness often have similar characteristics.

In accordance with the present invention, the NLGC part of the hearing aid, with its speech-noise discrimination operation, may be of the form illustrated in FIG. 2 for operation, for example, in Section 4 of FIG. 1. In FIG. 2, when the input signal envelope is between 50 to 60 dB or greater, Gate 1 remains closed and these time segments of the signal pass directly through towards the output, so-labelled, without the emphasis or extra gain available from amplifier B. When the signal envelope falls to a value indicating that the input signal is below 50 dB, Gate 1 opens and the signal in Path B (which has been amplified by amplifier B by a given amount relative to the level in Path A) is added to the signal present in Path A, provided that Gate 2 is also open. Gate 2, by means of the attack and release time control 2' is open when the signal envelope is more than the illustrative 50 dB; however, when the signal remains below 50 dB for more than 0.5 sec., Gate 2 closes, thereby preventing further gain-emphasized signal segments coming through Gate 1 from reaching Path A. Accordingly, the extra emphasis or amplification given to the signal by amplifier B is not added to the signal in Path A. Amplifier B is adjustable so that the amount of extra emphasis given to the signal, relative to its level in Path A, can be varied to best meet the needs of different degrees of hearing loss.

Rectifier R, amplifier 1'' and attack-release time control elements 1' and 2' perform the following functions: rectifier R provides a means of making the negative parts of the signal continuum positive in voltage; and amplifier 1'' is adjustable and provides a means for adjusting the rectified signal continuum level reaching the attack-release time controls 1' and 2'. Accordingly, depending on the desires of the user during a given input signal-noise condition, the signal continuum level can be increased or decreased from rectifier, R, so that the attack-release controls 1' and 2', which affect Gates 1 and 2, respectively, and which are set to operate at specified voltages, will be activated with different signal-continuum levels at the input to the microphones. Thus, amplifier 1'' provides a means of causing the gates to be activated with lesser or greater input signals at the microphone as will be advantageous to persons with different degrees of sensori-neural deafness.

The purpose of the described double-gate action is to give the weaker, short duration (less than 0.5 sec) segments of the signal the extra amplification or emphasis relative to the strong intensity segments of the speech signal; but not to give this extra amplification to relatively low intensity background noise which is typically more steady-state than the speech signal. This background noise may continue at a level below, say, 50 dB for much longer duration than 0.5 sec. and is especially objectionable to persons using hearing aids that provide automatic gain compression that typically increases the relative intensity of this background noise.

It is to be understood that for some types of speech or other signals, the attack and release times for the operation of Gates 1 and 2 may be changed for optimum results from those specified in FIG. 2. It is also to be noted that the NLGC processing system herein described has other possible applications beyond that in hearing aids where it is desirable to provide relative emphasis or de-emphasis to different segments of electronic signals that dynamically vary in intensity in somewhat predictable ways such that its use, while particularly adapted to the present invention, is also applicable in other signal processing systems wherein similar advantages are sought.

Suitable components for the circuit elements are as follows: Gates 1 and 2 may, for example, be of the Field Effect Transistor (FET) type, described in *Electronic Principles* by Malvino, McGraw-Hill, New York, 1973: attack and release time control circuits 1' and 2' may be of the operational amplifier type with appropriate capacitive and resistive feedback elements, as described in the same publication. Clearly, other types of well-known circuits may be similarly employed, and further modifications, within the spirit and scope of the invention, will suggest themselves to those skilled in this art.

What is claimed is:

1. A method of aiding hearing, that comprises, adjusting the over-all intensity level of speech signals with substantially linear gain over a predetermined range of intensities; applying the adjusted-intensity signals along a plurality of frequency filtering paths, one passing a broad band of the speech signal frequencies, and the others passing successive adjacent narrow bands within said broad band; reducing separately in each of the other paths, the dynamic range of intensity levels corresponding to segments of speech signals that vary in intensity for brief moments in the corresponding narrow bands, as distinguished from the more steady state segments of background noise and steady-state signals; and combining the signals from said paths.

2. A method as claimed in claim 1 and in which the signals in each of said paths are split and fed along a pair of further paths for right and left ear excitation, with the signal combining step being effectd by combining the right ear further paths and separately combining the left ear further paths.

3. A method as claimed in claim 2 and in which independent level adjustments are effected in each of the further paths prior to such combining.

4. A method as claimed in claim 1 and in which the speech signals are derived from a pair of right and left ear acoustic pick-up regions and a further pick-up region adjustable closer to the source of speech, and then the same are combined prior to said over-all intensity level adjusting step.

5. Hearing aid apparatus having, in combination, microphone pick-up means; automatic gain control means connected with the pick-up means to adjust the overall signal intensity level of speech signals with substantially linear gain over a predetermined range of intensities; a plurality of filter paths connected with the automatic gain control means and comprising a first path with broad band filter means for the speech signal frequencies and a plurality of further paths containing band-pass filters of successive adjacent narrow bands within said broad band; a plurality of speech-noise discrimination means, one connected in each of the plurality of further paths for separately reducing in each path the dynamic range of signal intensity levels corresponding to segments of speech signals that vary in intensity for brief moments in the respective narrow

bands, as distinguished from the more steady state segments of background noise and steady-state signals; and means for combining the signals from said paths.

6. Apparatus as claimed in claim **5** and in which said combining means comprises pairs of right and left ear paths, each pair split from the output of the broad band filter means and the outputs of the plurality of speech-noise discrimination means, with all right ear paths connected together and all left ear paths connected together.

7. Apparatus as claimed in claim **6** and in which said pairs of paths comprise separate variable gain amplifier means and resistive combining networks.

8. Apparatus as claimed in claim **7** and in which further variable gain amplifier means is provided at the output of each of the connected-together right and left ear paths, independently operable with respect to the said separate variable gain amplifier means.

9. Apparatus as claimed in claim **4** and in which each of said speech-noise discrimination means comprises a pair of signal processing paths connected to the corresponding band pass filter means, one of said paths including gating means and attack-release time control means for operating the gating means to apply amplification emphasis along the other processing path for the weaker short-duration segments of the signal relative to the strong intensity segments of the speech signal, but without providing added amplification to relatively low intensity background noise.

10. Apparatus as claimed in claim **4** and in which said microphone pick-up means comprises right and left ear microphones and a remote microphone adjustable to regions closer to the source of speech, with all of the microphones connected to the automatic gain control means.

11. Hearing aid apparatus having, in combination, right and left ear microphone pick-up means, remote microphone means adjustable to regions closer to the source of sound, and common automatic gain control means connected to all of said microphone means to receive the combined inputs thereof.

12. Hearing aid apparatus as claimed in claim **5** and in which frequency shift means is provided connected with the automatic gain control means for shifting the frequency of signals picked up by the microphone pick-up means and mechanical vibratory linkages associated therewith, said frequency shift means comprising means for modulating with one frequency and demodulating with a second and different frequency.

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