(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 327 617 B1**

(12) EUROPEAN PATENT SPECIFICATION

(45) Date of publication of patent specification: **04.11.92**

(51) Int. Cl.5: **G10K 11/16**, H04R 3/00

(21) Application number: **88905991.1**

(22) Date of filing: **20.07.88**

(86) International application number:
PCT/GB88/00581

(87) International publication number:
WO 89/00746 (26.01.89 89/03)

(54) IMPROVEMENTS RELATING TO NOISE REDUCTION SYSTEMS.

(30) Priority: **20.07.87 GB 8717043**
20.07.87 GB 8717040
20.07.87 GB 8717042

(43) Date of publication of application:
16.08.89 Bulletin 89/33

(45) Publication of the grant of the patent:
04.11.92 Bulletin 92/45

(84) Designated Contracting States:
AT BE CH DE FR GB IT LI LU NL SE

(56) References cited:
EP-A- 0 212 840
DE-A- 3 133 107
GB-A- 2 172 769
US-A- 4 061 875

(73) Proprietor: **Siemens Plessey Electronic Systems Limited**
Oakcroft Road
Chessington, Surrey KT9 1OZ(GB)

(72) Inventor: **TWINEY, Robert, Christopher**
7 Winchester Close Delapre
Northampton NN4 9BA(GB)
Inventor: **SALLOWAY, Anthony, John**
11 Furber Court Arbours
Northampton NN3 3RW(GB)

(74) Representative: **Allen, Derek**
Siemens Group Services Limited, Intellectual Property Department, Roke Manor, Old Salisbury Lane
Romsey, Hampshire SO51 0ZN(GB)

EP 0 327 617 B1

Description

This invention relates to systems for reducing the level of acoustic noise fields within ear-defenders or earphone structures worn by personnel (e.g., pilots, vehicle drivers, military personnel) in high noise environments.

Known active noise reduction (ANR) systems for reducing the acoustic noise field in ear-defenders comprise noise pick-up microphones and noise-cancelling sound generators (usually known as loudspeakers) mounted within the internal cavities or enclosures of the respective ear-defenders. The noise pick-up microphones produce electrical signal outputs in response to the acoustic noise fields within the cavities and these signal outputs are phase inverted, filtered and amplified in a feedback loop and fed to the noise-cancelling sound generators which produce noise-cancelling acoustic signals of substantially the same amplitude but of opposite phase to the acoustic noise field waveforms. The design considerations underlying such ANR systems are described in "Some transducer design considerations for earphone active noise reduction systems", Twiney et al., Vol. 7 part 2. pp.95-102, Proc. Spring Conference, 1985, York, Institute of Acoustics. A specific arrangement of an ANR-system is disclosed, for example, in EP-A-0 212 840.

Problems arise through inherent imperfections in the pick-up microphones and sound generators, by way of unwanted phase changes producing signal enhancement or by way of failure to cope with large amplitude signals in certain frequency regions.

One problem which occurs is that of large pressure pulses (buffets) which occur inside an ear-defender or earphone structure due to relative movement between the human head and the earphone, or propagate to the earphone from a device that causes a rapid pressure change, e.g. a gun, helicopter, vehicle, explosive device. These pulses are very high in amplitude, and create large signals in the feedback loop as a result of high system loop gain. Due to the inadequacy of the sound generator to produce enough sound output, drive voltages appear at the sound generators which are higher than the maximum input voltage, and may overdrive the sound generator and cause permanent failure.

Another problem which arises is that of signal enhancement at certain frequencies within the bandwidth of the feedback loop wherein due to imperfect transfer functions of the noise pickup microphone and sound generator the ANR will, at certain frequencies be feeding in-phase(i.e. positive feedback) signals rather than anti-phase(i.e. negative feedback) signals to the sound generator.

A further problem which occurs is that due to the imperfect transfer functions of both the microphone and generator, the total bandwidth for feedback signals having an appropriate phase is limited, being bounded by regions in which positive feedback occurs. It is usual to employ in feedback systems in general a lowpass first order filter operating at a high frequency in order to stabilize the loop. However such first order low pass filters are not appropriate for filtering out sound energy frequencies in ANR systems because of the large phase changes which occur in the cut-off regions which give rise to problems of positive feedback and signal enhancement.

It was previously thought, as appears from the article referred to above, that electronic processing to overcome problems in ANR systems had limited application because of the causal relation between amplitude and phase response of electronic filters.

Nevertheless it has now been found as a result of careful investigation into the problems arising in feedback loops of ANR systems, that electronic processing may be used to advantage.

It is an object of the present invention to overcome one or more of the above problems.

According to the present invention there is provided an active noise reduction system comprising a noise-cancelling sound generator, a microphone acoustically coupled to said generator, a feedback loop connected between said microphone and said generator, and including loop stabilisation means for inverting the phase of microphone signals and filtering the microphone signals, means for amplifying the phase inverted and filtered signals, and a first high pass frequency filter means for filtering out sound energy from high pressure sound pulses arising from low frequency buffets, characterised in that the first filter means has a gain characteristic which is relatively low in a first frequency band and then rises continuously to a relatively high value in a second frequency band, said first filter means introducing a phase shift which rises to a maximum value in a transitional region of said gain characteristic of said first high pass frequency filter means, said system further comprising a second high pass filter means coupled between the loop stabilising means and the amplifying means for increasing loop gain and/or adjusting phase shift by predetermined amounts within one or more predetermined frequency bands, wherein the second filter means has a gain characteristic which is relatively low in a first frequency band and then rises continuously to a relatively high value in a second frequency band, the phase shift introduced by the second filter means rises to a maximum value in the transitional region of filter gain characteristic.

Thus this aspect of the invention is based on the recognition that the major part of sound energy in high pressure pulses is present at low frequencies say below 100 Hz and thus the provision of low frequency filter means in the feedback loop can reduce a major part of the sound energy in the pulses. Such filter means is conveniently referred to as an anti-buffet filter (ABF). The amount of ABF correction is limited because stability of the feedback loop must be maintained, the total loop gain being kept below unity where the total phase shift may cause constructive interference.

The filter means referred to above may be used in conjunction with a voltage limiting means, which prevents the generator from being overdriven by amplification of high pressure sound pulses. Such voltage limiting means may comprise a nonlinear amplifier or zener diode arrangement.

Thus in accordance with the invention, the provision of second filter means increasing gain or adjusting phase shift, preferably both, prevents enhancement of the signal in the feedback loop arising from imperfect transfer functions of the microphone and generator. The further filter means is conveniently termed an anti-enhancement filter (AEF). The amount of AEF correction is limited by the need to maintain stability of the feedback loop (the total loop gain must be kept below unity when the total phase shift may cause constructive interference).

This further aspect of the invention is based on our discovery that enhancement problems caused by transducer imperfections arise in a frequency region centred at about 500 Hz where the gain decreases whereas the phase lag in this area increases to about $3\pi/2$. Thus a high pass filter which adjusts the gain in this region whilst providing a phase advance compensating phase shift can significantly reduce the problems of signal enhancement.

In a particularly preferred form of the invention, it has been discovered as a result of careful investigation into the operability of ANR systems that the functions of the ABF and AEF, which operate at different frequencies and with different transfer functions can be accomplished by the use of a single high pass filter (ABEF) for attenuating frequencies below a predetermined frequency, the ABEF having appropriate transfer characteristics to prevent phase shifts harmful to loop stability.

The preferred speech signals are injected at a single point in the feedback loop between the AEF and the amplifying means, in order that the speech signals are substantially uncoloured by the AEF and other filters. It will be understood that the speech signals are in a frequency range which is for the most part above the frequency range in which the ANR is operative and the speech signals are not therefore reduced. They may however be affected by higher frequency filters in the feedback loop.

A preferred embodiment of the invention will now be described with reference to the accompanying drawing wherein:-

Figure 1 is a schematic diagram of an active noise reduction system according to the present invention;

Figure 2 is a circuit diagram of a preferred anti-buffet (ABF)filter;

Figure 3 is a graph of the ABF characteristics;

Figure 4 is a circuit diagram of a preferred anti-enhancement filter (AEF), and Figure 5 is a graph of the filter characteristics;

Figure 6 is a circuit diagram of a low pass filter for defining an upper limit of the feedback loop bandwidth;

Figure 7 is a graph of the low pass filter characteristics;

Figure 8 is a circuit diagram of an ABEF combining both AEF and ABF characteristics;

Figure 9 is a graph of the transfer functions of the ABEF of figure 8.

Referring to Figure 1 of the Drawings, the active noise reduction system illustrated comprises a generally cup-shaped circumaural earphone structure 1 arranged to enclose the wearer's ear 2. The rim of the structure 1 is cushioned against the side of the wearer's head 3 by means of a compliant ring cushion 4. The earphone structure 1 embodies a small noise pick-up microphone 5, which detects the noise within the earphone adjacent to the wearer's ear 2 and provides an electrical output dependent upon the detected noise. This output signal from the microphone is passed through an anti-buffet filter 6, a loop stabilisation unit 7,a low-pass filter 8, an anti-enhancement filter 10 and amplifier 12, to a noise cancelling sound generator (loudspeaker) 14 which is mounted on a baffle 16 within structure 1. Loop stabilisation unit 7 includes a phase inverter 72, a loop stabilizing filter 74 (which may be incorporated in low-pass filter 8 as in Figure 6) for filtering out very high frequencies, and a voltage limiting circuit 76 comprising a zener diode switching arrangement for limiting high amplitude input signals. Filter 6 is placed first in the feedback loop in order to minimise signal values in the loop. The effect of the anti-enhancement filter is to reduce noise effects arising from imperfect transfer functions of microphone 5 and generator 14.

A speech signal is injected between anti-enhancement filter 10 and amplifier 12 at an input node 18. The introduction of the speech signal at this point allows the speech signal to be substantially uncoloured by the loop filters. If desired the speech signals may be pre-emphasised by amplification where they may be attenuated by the

ANR system.

Referring to Figure 2, the ABF 6 comprises an amplifier 20 having a negative feedback loop with a resistor R1 connected to its inverting input, which receives an input signal from a resistive/capacitive network R2, R3, C1. The non-inverting input of the amplifier is connected through a resistor R4 to ground.

The characteristics of ABF 6 are shown in Figure 3, whence it may be seen that the filter has a loss factor of about 8db up to about 100 Hz at which frequency the loss reduces continuously until at about 500Hz the filter exhibits a small gain factor.

The phase shift introduced by the filter is an advance with increasing frequency rising in the transitional region from the base level of substantially 180° (the filter includes an inverting amplifier) to a maximum at about 200 Hz of about 215°. This phase shift must be taken into account when considering the overall loop stability. The effect of the ABF 6 on the overall feedback loop transfer function is to attenuate the low frequency end of the function whereby noise in the frequency range up to 200Hz is severely attenuated.

The preferred form of AEF is shown in Figure 4 as comprising two cascaded stages 21, 22, each stage comprising an amplifier 24 with a resistor R1 in a negative feedback loop and with the inverting amplifier input being connected to ground via the series combination of a resistor R2 and capacitor C1. The filter characteristics are shown in Figure 5 with the gain having step-form, being roughly 0db up to 100 Hz and then rising to 10db gain at 1 kHz. The phase shift, a phase advance with increasing frequency, rises in the region in which the gain changes, from a base level of substantially 0° a maximum value of 25° at roughly 500 Hz.

Because of the precise transfer functions of the microphone and generator, the gain reduces to a minimum value at about 500 Hz whereas the phase shift in this area rises to a maximum of about more than $3\pi/2$. By providing AEF, the transfer functions are modified in this area to reduce phase shift and increase gain, thereby reducing signal enhancement.

A circuit diagram of low pass filter 8 is shown in Figure 6 as comprising a transitional second order filter including an amplifier 60 having a non-inverting input connected to a filter input via resistors R1, R2 and a capacitor C1 coupled between the amplifier input and ground. Two feedback loops are provided from the amplifier output to the non-inverting input: a first loop including a capacitor C2 and a second loop comprising resistors R3, R4, R5 and a capacitor C3 in series with a resistor R9 connected between resistors R4, R5 and ground. A further feedback loop is provided comprising a resistor R7 connected between the amplifier output and the inverting amplifier input. A further resistor R8 is connected between resistor R7 and ground.

The characteristics of the filter are shown in Figure 7 where the gain is close to 0db up to about 1000Hz and is about -30db around 10,000Hz. The gain decreases between these regions relatively quickly in a cut-off region.

The phase shift across the filter is roughly 150°, (the filter includes a non-inverting amplifier) in the region below 1,000Hz and above 10,000Hz, but decreases to a minimum, (a phase lag with increasing frequency) of about 45° in the centre of the cut-off region. Such a phase change of roughly 105° is acceptable and is much smaller than 180 degrees resulting from a conventional second order low-pass filter. Although a second order filter is shown, the filter could be a higher or lower order if desired.

Referring now to Figure 8, there is shown a high pass filter which combines the functions of the AEF and ABF and is herein referred to as an ABEF. The filter is a second order filter comprising two filter sections connected in cascade, the filter sections being identical. (If desired a first order filter could be employed). Each filter section comprises an input port 80 coupled to the inverting input of an amplifier 82 through a resistance R1 connected in parallel with a capacitance C1 and a resistance R2. The non-inverting input of the amplifier is connected to ground via a resistance R3, and the output of the amplifier 86 is connected in a negative feedback loop to the inverting input of the amplifier via a resistor R4.

Referring now to Figure 9, the characteristics of the filter of Figure 8 are shown where the gain is slightly greater than 0db up to about 500 Hz and then rises to about 10db at a frequency of 2 kHz in a transitional region between 500Hz-2kHz. The phase shift changes from a constant level of about 0° to a maximum value of substantially 60° at about 1 kHz.

It will be appreciated that although the particular embodiment specifically described is applied to a circumaural earphone structure, the invention may also be applied to other earphone structures such as the supra-aural type.

It will also be appreciated that the filters shown may be replaced by digital filters, and the elements of the feedback loop may be digitised by employing a micro-computer with appropriate routines. The invention claimed is intended to cover both analog and digital systems.

Claims

1. An active noise reduction system comprising a noise-cancelling sound generator (14), a micro-

phone (5) acoustically coupled to said generator (14), a feedback loop connected between said microphone (5) and said generator (14), and including loop stabilisation means (7) for inverting the phase of microphone signals and filtering the microphone signals, means (12) for amplifying the phase inverted and filtered signals, and a first high pass frequency filter means (6) for filtering out sound energy from high pressure sound pulses arising from low frequency buffets, characterised in that the first filter means (6) has a gain characteristic which is relatively low in a first frequency band and then rises continuously to a relatively high value in a second frequency band, said first filter means (6) introducing a phase shift which rises to a maximum value in a transitional region of said gain characteristic of said first high pass frequency filter means, said system further comprising a second high pass filter means (10) coupled between the loop stabilising means (7) and the amplifying means (12) for increasing loop gain and/or adjusting phase shift by predetermined amounts within one or more predetermined frequency bands, wherein the second filter means (10) has a gain characteristic which is relatively low in a first frequency band and then rises continuously to a relatively high value in a second frequency band, the phase shift introduced by the second filter means (10) rises to a maximum value in the transitional region of filter gain characteristic.

2. A system as claimed in claim 1, wherein the second filter means (10) includes one or more stages, each stage comprising an amplifier (24) with a negative resistive feedback loop (20, 22), and a resistive capacitive path (R2, C1) to a reference potential connected to the inverting amplifier input.

3. A system as claimed in claim 1 or claim 2, wherein the functions of the first (6) and second (10) filters are provided by a single high pass filter means, with the transfer characteristics selected to filter out sound energy from high pressure sound pulses and to increase loop gain and/or adjust phase shift to compensate for transfer characteristics of the microphone (5) and generator (10).

4. A system as claimed in any preceding claim, and further comprising means for injecting a speech signal into the feedback loop between the filter means (10) and the amplifying means (12).

5. A system as claimed in any preceding claim, wherein the second high pass filter means (10) includes one or more stages, each stage including a first order high pass active filter, the active filter network being coupled to the operational amplifier for providing a high pass characteristic.

6. A system as claimed in claim 5, wherein the first filter means (6) comprises one or more stages, each stage comprising a first order high pass active filter.

7. A system as claimed in claim 6, wherein the first high pass frequency filter means (6) comprises a resistive-capacitive combination (R2, R3, C1) and an amplifier (20) having inverting and non-inverting inputs with a negative feedback loop (R1), an input of the filter means being provided on one side of said resistive-capacitive combination (R2, R3, C1) and the other side of said combination being connected to said inverting input and the non-inverting input being connected to reference potential via a resistive connection (R4).

8. A system as claimed in any preceding claim, wherein said loop stabilisation means (7) includes means (76) for limiting the amplitude of the signals in the feedback loop.

9. A system as claimed in any preceding claim, wherein the first high pass frequency filter means (6) is connected between the microphone (5) and said loop stabilisation means (7).

Patentansprüche

1. Anordnung zur aktiven Störschallverminderung mit einem störschallunterdrückenden Tongenerator (14), einem mit dem Generator (14) akustisch gekoppelten Mikrofon (5), einem zwischen dem Mikrofon (5) und Generator (14) angeordneten Rückkoppelungskreis, und mit Rückkoppelungs-Stabilisierungsmitteln (7), um die Phase der Mikrofonsignale zu invertieren und die Mikrofonsignale zu filtern, Mitteln (12) zur Verstärkung der phaseninvertierten und gefilterten Signale, und einem ersten Hochpass-Frequenzfilter (6), um die Schallenergie von Hochdruck-Schallimpulsen herauszufiltern, die von tiefen Frequenzstössen herrühren, dadurch gekennzeichnet, dass das erste Filter (6) eine Verstärkungscharakteristik besitzt, welche in einem ersten Frequenzband verhältnismässig tief ist und danach kontinuierlich zu einem verhältnismässig hohen Wert in einem zweiten

Frequenzband ansteigt, wobei das erste Filter (6) eine Phasenverschiebung einleitet, welche zu einem Höchstwert in einem Übergangsbereich der Verstärkungscharakteristik des ersten Hochpass-Frequenzfilters ansteigt, und die Anordnung weiter ein zweites, zwischen dem Rückkoppelungs-Stabilisierungsmittel (7) und dem Verstärker (12) angeordnetes Hochpass-Filter (10) für die Verstärkung der Verstärkung der Rückkoppelung und/oder Einstellung der Phasenverschiebung durch vorbestimmte Werte innerhalb eines oder mehrerer Frequenzbänder aufweist, worin das zweite Filter (10) eine Verstärkungscharakteristik besitzt, welche in einem ersten Frequenzband verhältnismässig tief ist und danach kontinuierlich zu einem verhältnismässig hohen Wert in einem zweiten Frequenzband ansteigt, wobei die durch das zweite Filter (10) eingeleitete Phasenverschiebung zu einem Höchstwert im Übergangsbereich der Verstärkungscharakteristik ansteigt.

2. Anordnung nach Anspruch 1, worin das zweite Filter (10) eine oder mehrere Stufen enthält, wobei jede Stufe einen Verstärker (24) mit einem negativen Widerstands-Rückkoppelungskreis (20,22) und eine Widerstands-Kapazitäts-Bahn (R2,C1) zu einem mit dem invertierenden Verstärkereingang verbundenen Referenzpotential aufweist.

3. Anordnung nach Anspruch 1 oder 2, worin die Funktionen des ersten (6) und zweiten (10) Filters durch ein einziges Hochpassfilter vorgesehen sind, welches die zur Ausfilterung von Schallenergie von Hochdruck-Schallimpulsen ausgewählte Übertragungscharakteristik aufweist und die Verstärkung der Rückkoppelung und/oder Einstellung der Phasenverschiebung verstärkt, um die Übertragungscharakteristik des Mikrofons (5) und des Generators (10) zu kompensieren.

4. Anordnung nach jedem der vorangegangenen Ansprüche, mit Mitteln für die Einführung eines Sprechsignales in den Rückkoppelungskreis zwischen dem Filter (10) und dem Verstärker (12).

5. Anordnung nach jedem der vorangegangenen Ansprüche, worin das zweite Hochpassfilter (10) eine oder mehrere Stufen beinhaltet, wobei jede Stufe ein aktives Hochpassfilter erster Ordnung beinhaltet, und das aktive Filternetzwerk mit dem Operationsverstärker verbunden ist, um eine Hochpass-Charakteristik vorzusehen.

6. Anordnung nach Anspruch 5, worin das erste Filter (6) eine oder mehrere Stufen beinhaltet, wobei jede Stufe ein aktives Hochpassfilter erster Ordnung beinhaltet.

7. Anordnung nach Anspruch 6, worin das erste Hochpass-Frequenzfilter (6) eine Widerstands-Kapazitäts-Kombination (R2,R3,C1) und einen Verstärker (20) mit invertierenden und nichtinvertierenden Eingängen mit einem negativen Rückkoppelungskreis (R1) aufweist, wobei ein Eingang des Filters auf einer Seite der Widerstands-Kapazitäts-Kombination (R2,R3,C1) vorgesehen ist und die andere Seite dieser Kombination mit dem invertierenden Eingang des Verstärkers verbunden ist und der nichtinvertierende Eingang über einen Widerstand (R4) mit dem Referenzpotential verbunden ist.

8. Anordnung nach jedem der vorangegangenen Ansprüche, worin die Rückkopplungs-Stabilisierungsmittel (7) Mittel (76) zur Begrenzung der Amplitude der Signale im Rückkoppelungskreis beinhalten.

9. Anordnung nach jedem der vorangegangenen Ansprüche, worin das erste Hochpass-Frequenzfilter (6) zwischen dem Mikrofon (5) und den Rückkoppelungs-Stabilisierungsmitteln (7) angeordnet ist.

Revendications

1. Système de réduction active de bruit, comprenant un générateur sonore (14) de compensation de bruit, un microphone (5) couplé de manière acoustique au générateur (14), une boucle de rétroaction connectée entre le microphone (5) et le générateur (14) et comprenant un dispositif (7) de stabilisation de boucle destiné à inverser la phase des signaux du microphone et à filtrer les signaux du microphone, un dispositif (12) destiné à amplifier les signaux ayant subi l'inversion de phase et le filtrage, et un premier filtre passe-haut (6) destiné à retirer l'énergie sonore des impulsions sonores à haute pression dues à des souffles à basses fréquences, caractérisé en ce que le premier dispositif de filtrage (6) a une caractéristique de gain qui est relativement faible dans une première bande de fréquences et qui s'élève de façon continue jusqu'à une valeur relativement élevée dans une seconde bande de fréquences, le premier dispositif de filtrage (6) introduisant un déphasage qui s'élève jusqu'à une valeur maximale dans une région de transition de la caractéristique du gain du premier dispositif de filtrage passe-haut, le système comprenant en outre un second dispositif de filtrage passe-haut (10) couplé entre le dispositif (7) de stabilisation de boucle et le dispositif amplificateur (12) afin qu'il augmente le gain de boucle et/ou ajuste le déphasage de quantités prédéterminées dans une ou plusieurs bandes prédéterminées de fréquences, dans lequel le second dispositif de filtrage (10) a une caractéristique de gain qui est relativement faible dans une première bande de fréquences et qui s'élève de façon continue à une valeur relativement élevée dans une seconde bande de fréquences, le déphasage introduit par le second dispositif de filtrage (10) augmentant jusqu'à une valeur maximale dans la région de transition des caractéristiques de gain du filtre.

2. Système selon la revendication 1, dans lequel le second dispositif de filtrage (10) comprend un ou plusieurs étages, chaque étage comprenant un amplificateur (24) avec une boucle (20, 22) de rétroaction résistive négative, et un trajet résistif et capacitif (R2, C1) relié à un potentiel de référence connecté à l'entrée d'inversion de l'amplificateur.

3. Système selon la revendication 1 ou 2, dans lequel les fonctions du premier (6) et du second (10) filtre sont assurées par un seul dispositif de filtrage passe-haut, les caractéristiques de transfert étant sélectionnées afin qu'elles retirent par filtrage l'énergie sonore des impulsions sonores à haute pression et augmentent le gain de boucle et/ou ajustent le déphasage pour compenser les caractéristiques de transfert du microphone (5) et du générateur (10).

4. Système selon l'une quelconque des revendications précédentes, comprenant en outre un dispositif d'injection d'un signal de parole dans la boucle de rétroaction entre le dispositif de filtrage (10) et le dispositif d'amplification (12).

5. Système selon l'une quelconque des revendications précédentes, dans lequel le second dispositif de filtrage passe-haut (10) comprend un ou plusieurs étages, chaque étage ayant un filtre actif passe-haut du premier ordre, le réseau actif de filtrage étant couplé à l'amplificateur opérationnel afin qu'il donne une caractéristique passe-haut.

6. Système selon la revendication 5, dans lequel le premier dispositif de filtrage (6) comporte un ou plusieurs étages, chaque étage comprenant un filtre actif passe-haut du premier ordre.

7. Système selon la revendication 6, dans lequel le premier dispositif (6) de filtrage passe-haut comprend une combinaison résistive et capacitive (R2, R3, C1) et un amplificateur (20) ayant des entrées d'inversion et de non-inversion avec une boucle de rétroaction négative (R1), une entrée du dispositif de filtrage étant placée d'un premier côté de la combinaison résistive et capacitive (R2, R3, C1) et l'autre côté de la combinaison étant connecté à l'entrée d'inversion et l'entrée de non-inversion étant connectée à un potentiel de référence par une connexion résistive (R4).

8. Système selon l'une quelconque des revendications précédentes, dans lequel un dispositif (7) de stabilisation de boucle comporte un dispositif (76) destiné à limiter l'amplitude des signaux dans la boucle de rétroaction.

9. Système selon l'une quelconque des revendications précédentes, dans lequel le premier dispositif (6) de filtrage passe-haut est connecté entre le microphone (5) et le dispositif (7) de stabilisation de boucle.

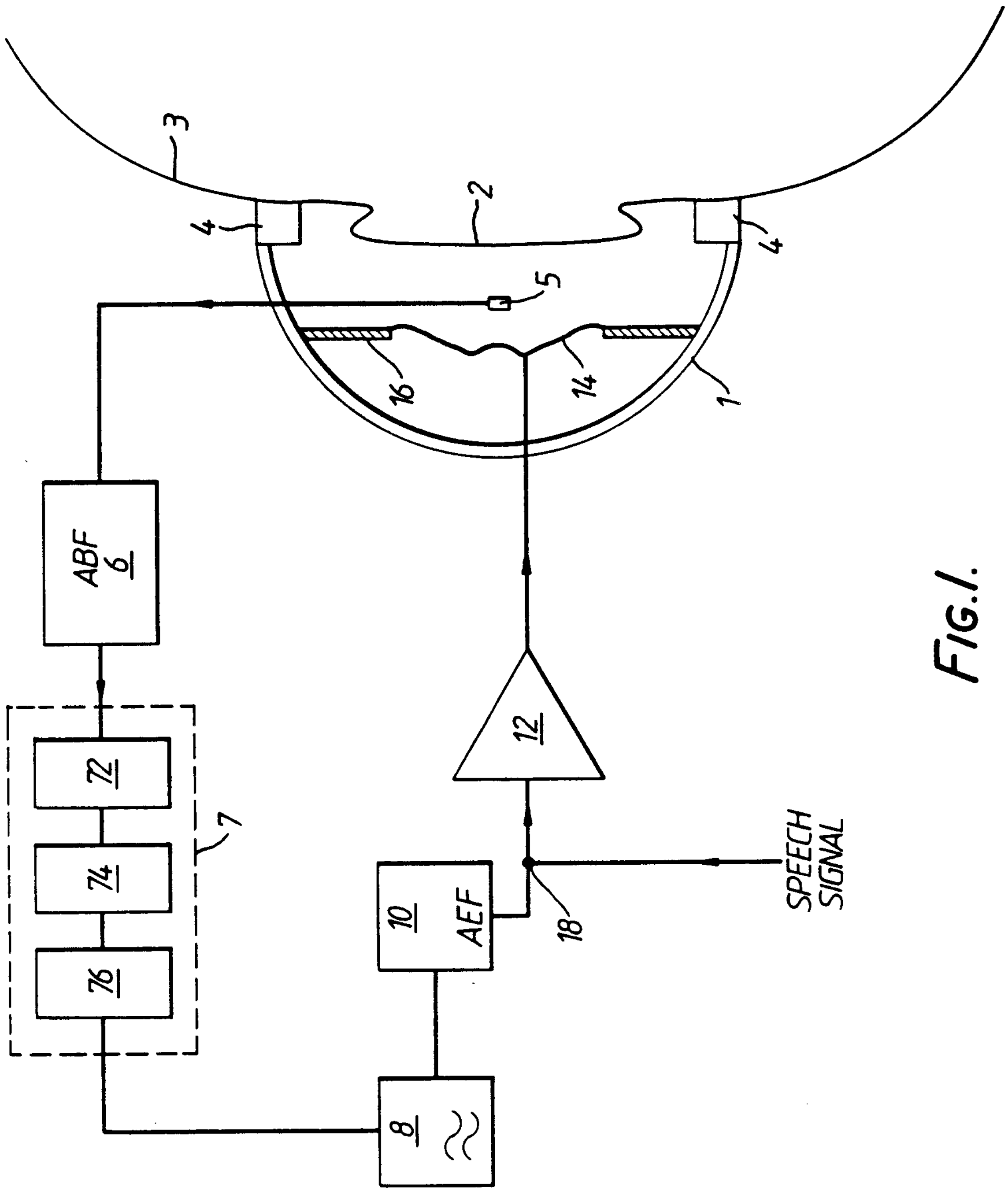

FIG.1.

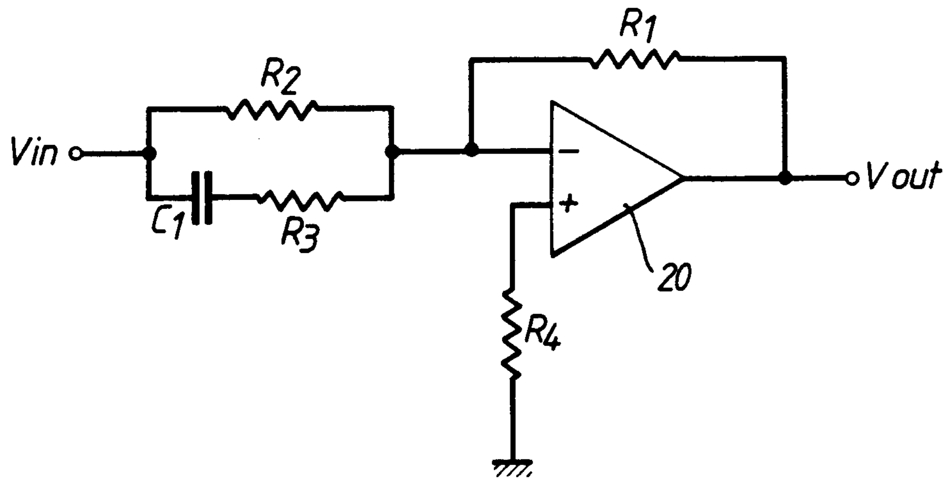

FIG. 2.

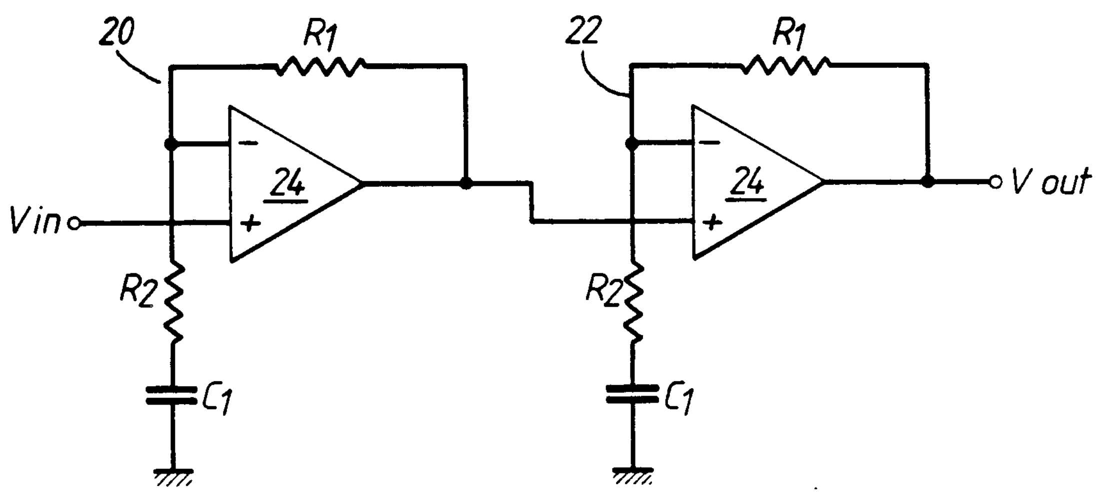

FIG. 4.

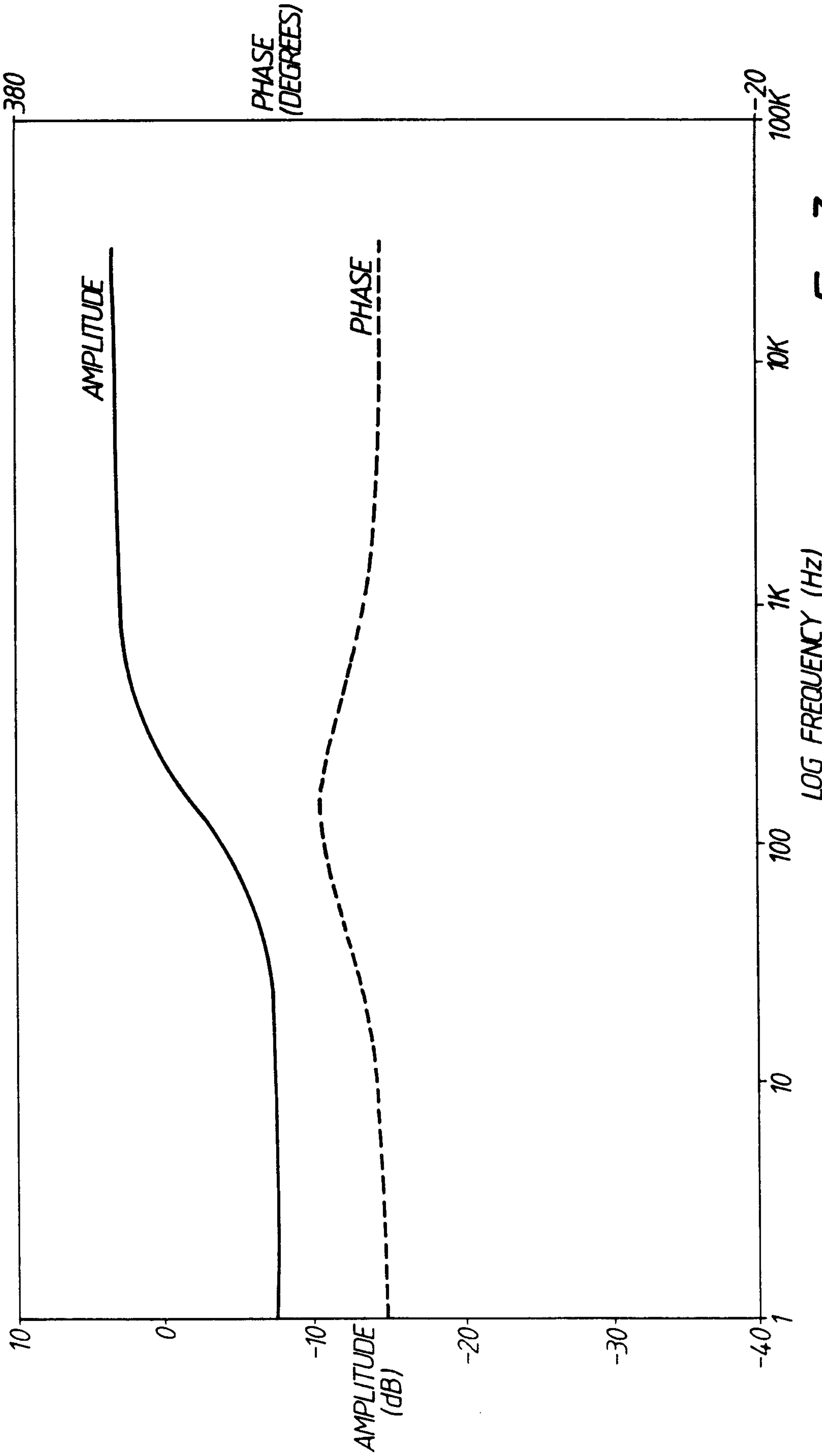
AMPLITUDE
PHASE
PHASE (DEGREES)
380
-20
AMPLITUDE (dB)
10
0
-10
-20
-30
-40
1
10
100
1K
10K
100K
LOG FREQUENCY (Hz)

FIG. 3.

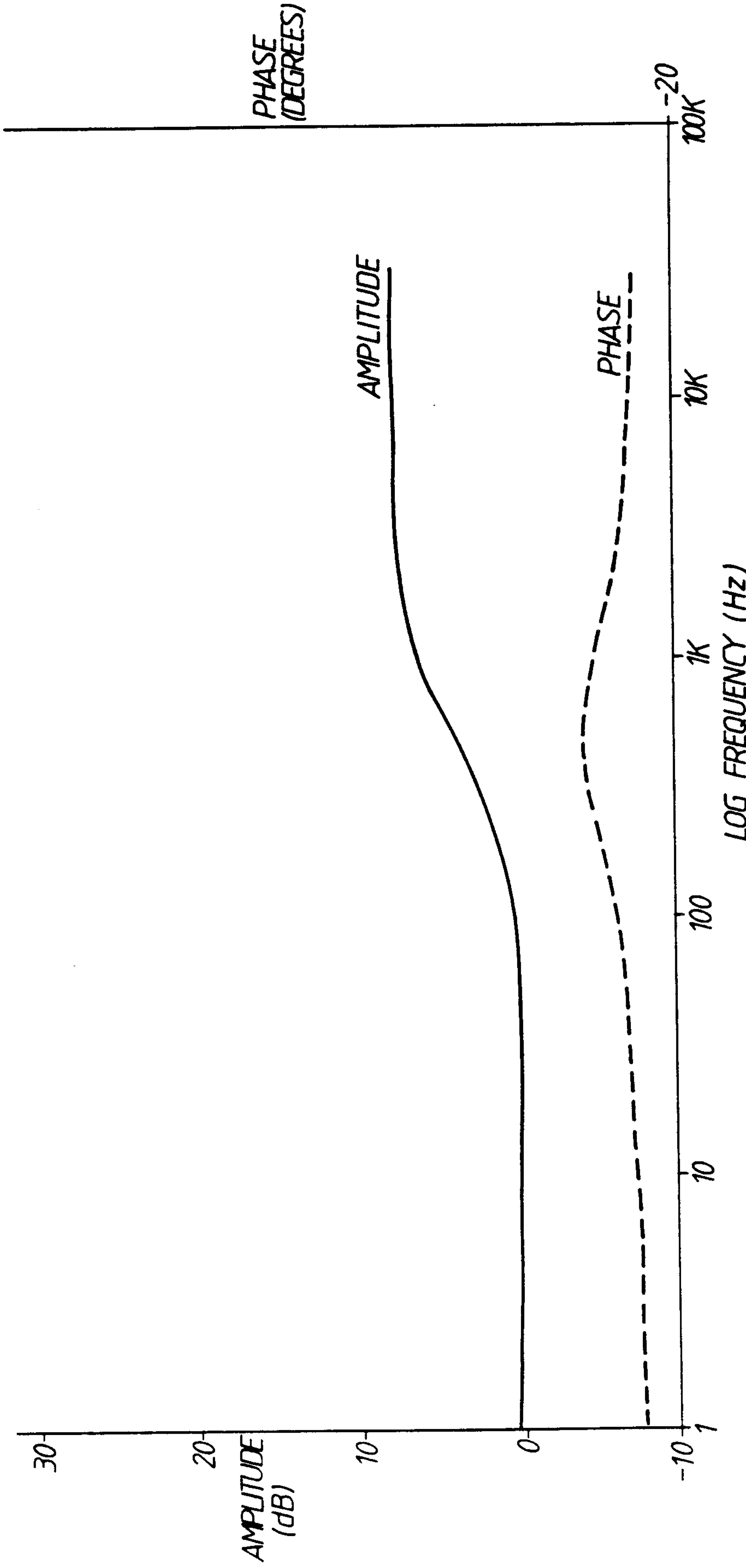
AMPLITUDE
(dB)
30
20
10
0
-10
1
10
100
1K
10K
100K
-20
PHASE
(DEGREES)
AMPLITUDE
PHASE
LOG FREQUENCY (Hz)

FIG. 5.

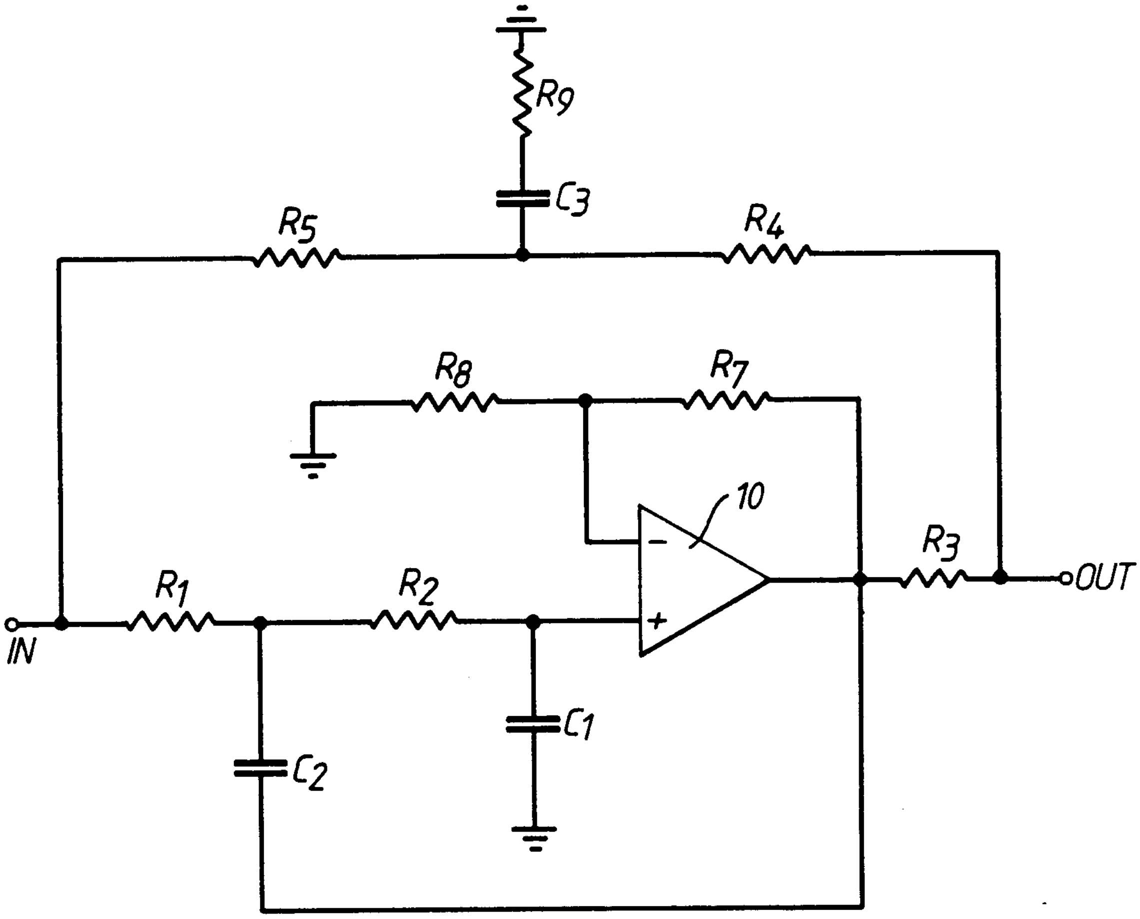
R9
C3
R5
R4
R8
R7
10
R3
OUT
R1
R2
IN
C1
C2

FIG.6.

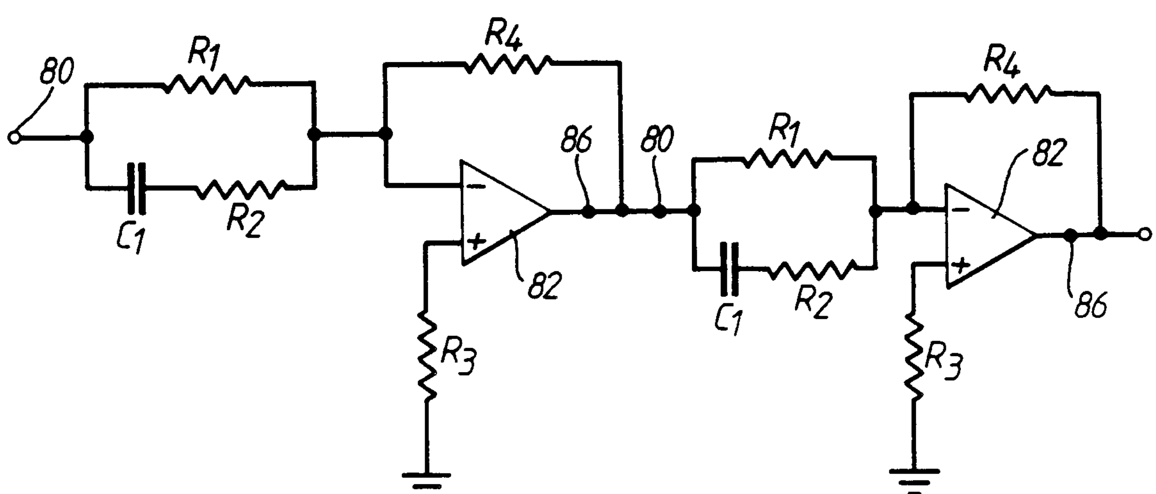
80
R1
R4
86
80
R1
R4
82
C1
R2
82
R3
C1
R2
86
R3

FIG.8.

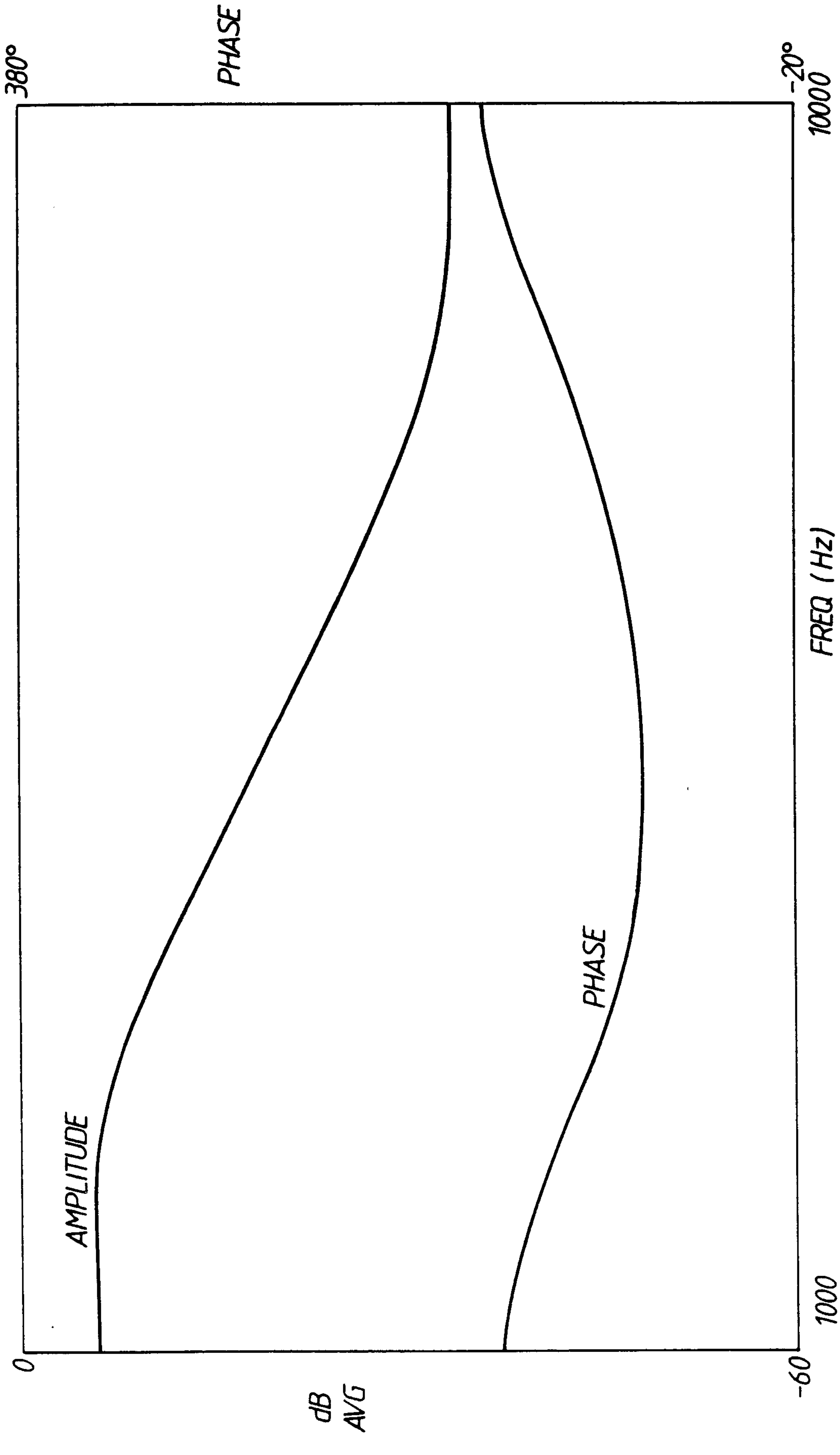

FIG.7.

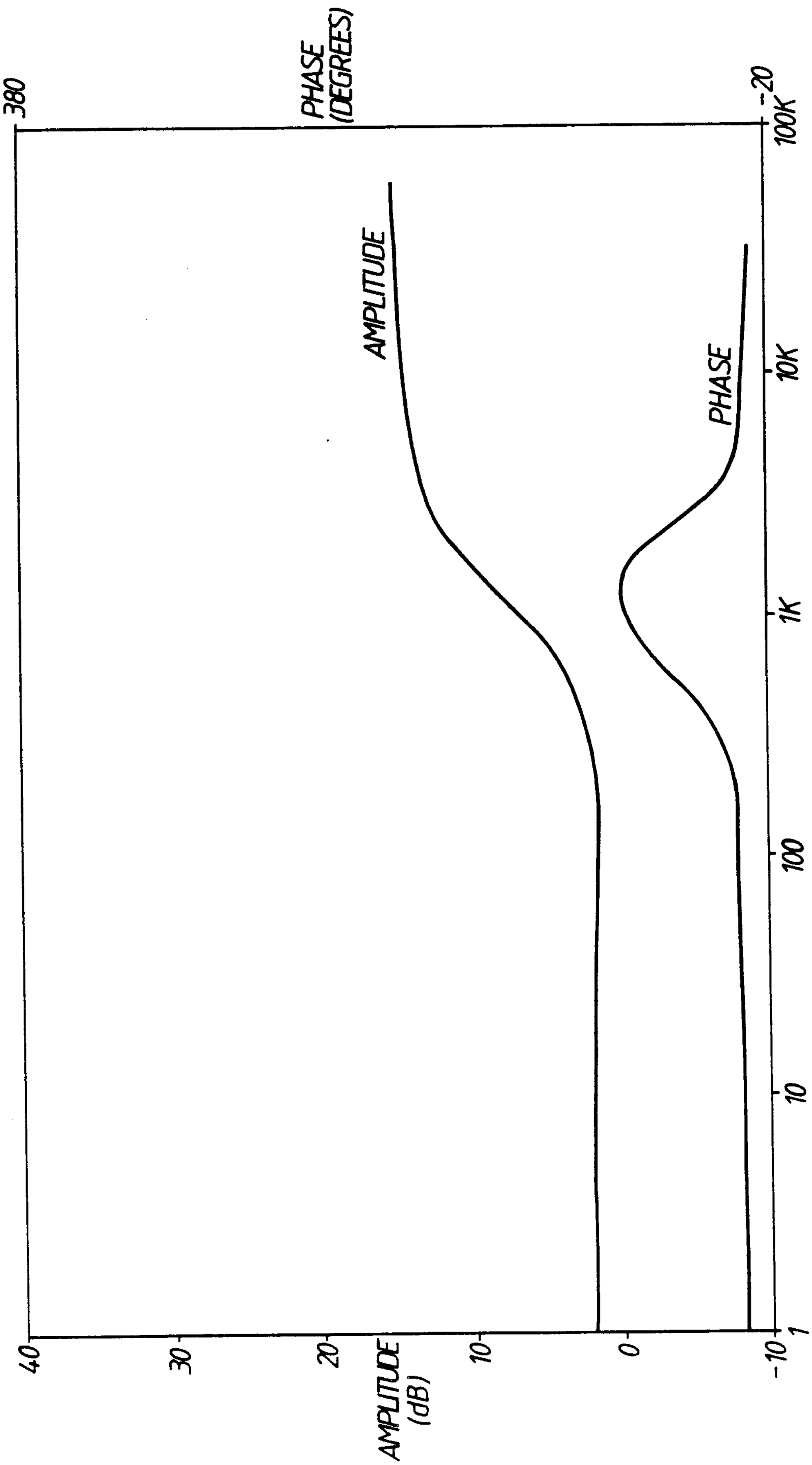
PHASE (DEGREES)
380
-20
AMPLITUDE
PHASE
AMPLITUDE (dB)
40
30
20
10
0
-10
1
10
100
1K
10K
100K
LOG FREQUENCY (Hz)

FIG.9.