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(56) Documents Cited:
GB 2293078 A
EP 2337376 A1
US 5406635 A
US 20100151787 A1
US 20050226444 A1
US 20050207567 A1

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(54) Title of the Invention: **Processing audio signals**
Abstract Title: **Reducing howling by applying a noise attenuation factor to a frequency which has above average gain**

(57) A method of reducing noise (howling) in a signal received at a processing stage of an acoustic system, the method comprising, at the processing stage: identifying at least one frequency at which a system gain of the acoustic system is above an average system gain of the acoustic system; providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and applying the noise attenuation factor to a component of the signal at that frequency.

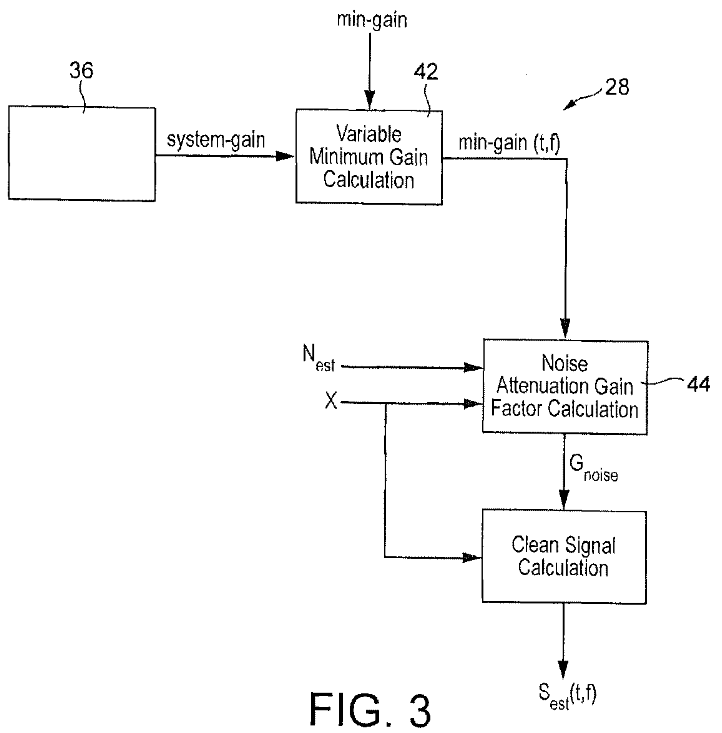

FIG. 3

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At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

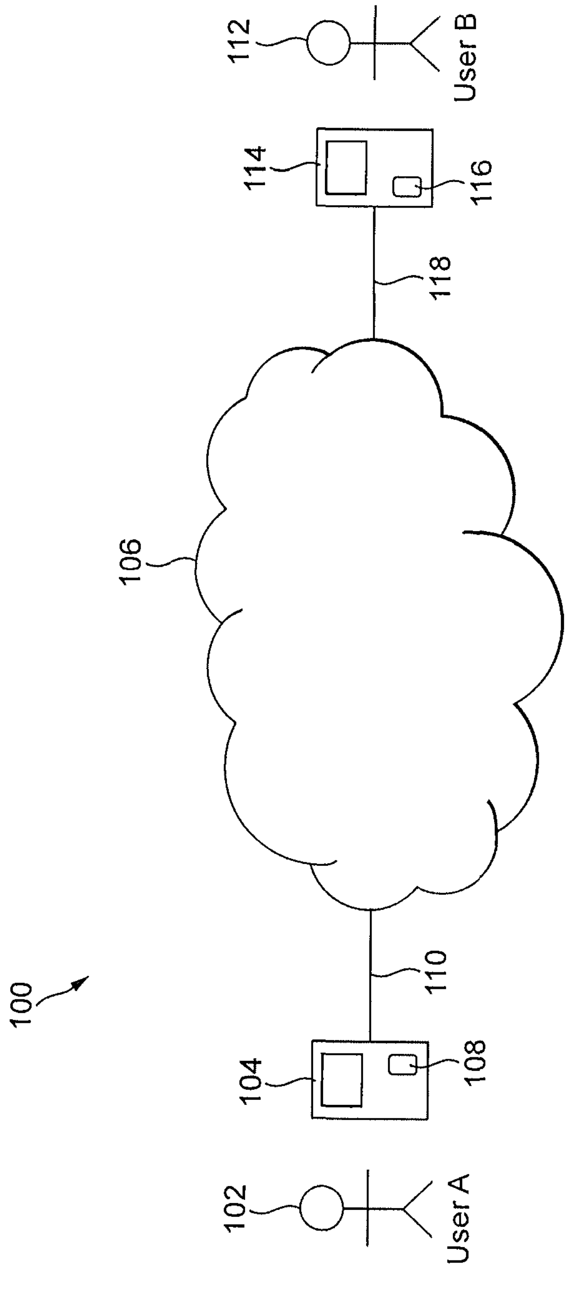

FIG. 1

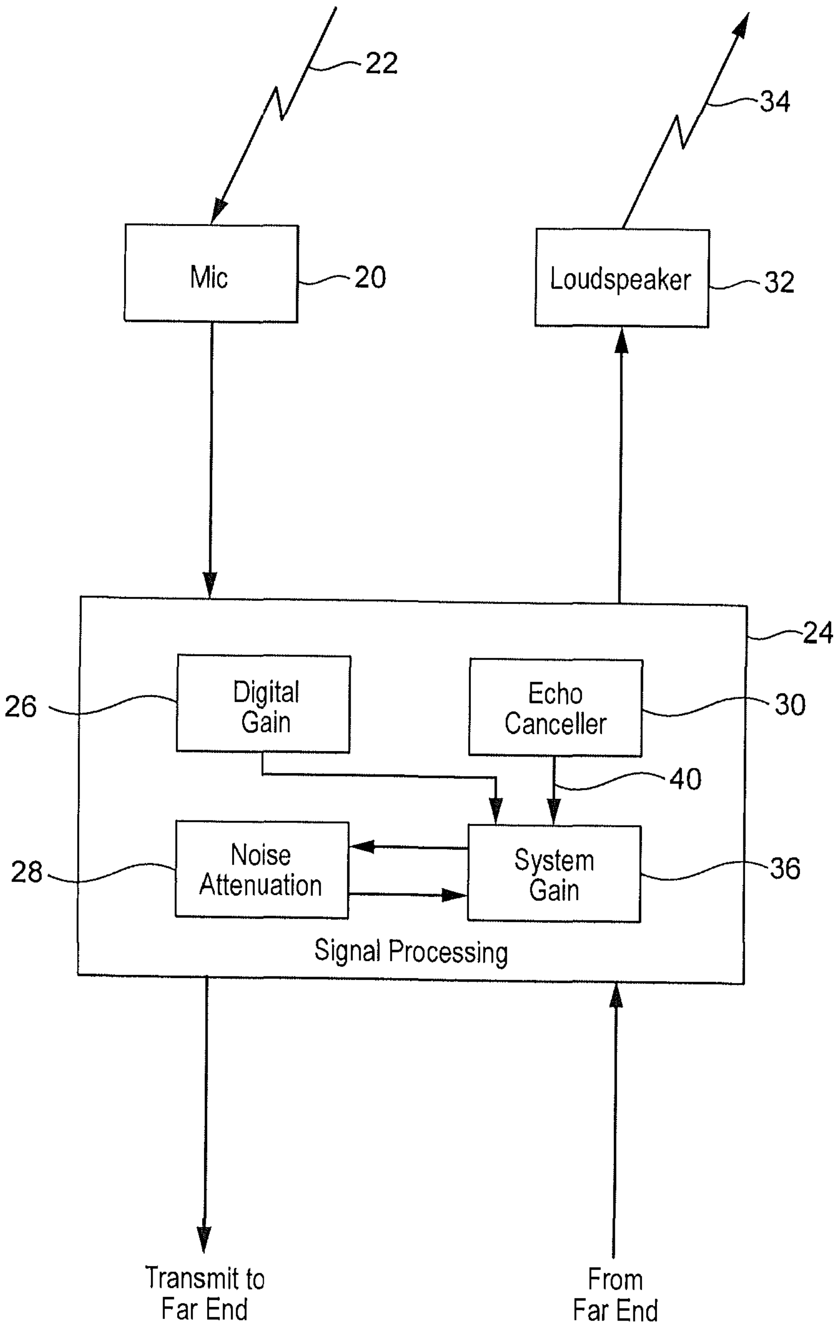
22
34
Mic
20
Loudspeaker
32
24
26
Digital Gain
Echo Canceller
30
40
28
Noise Attenuation
System Gain
36
Signal Processing
Transmit to Far End
From Far End

FIG. 2

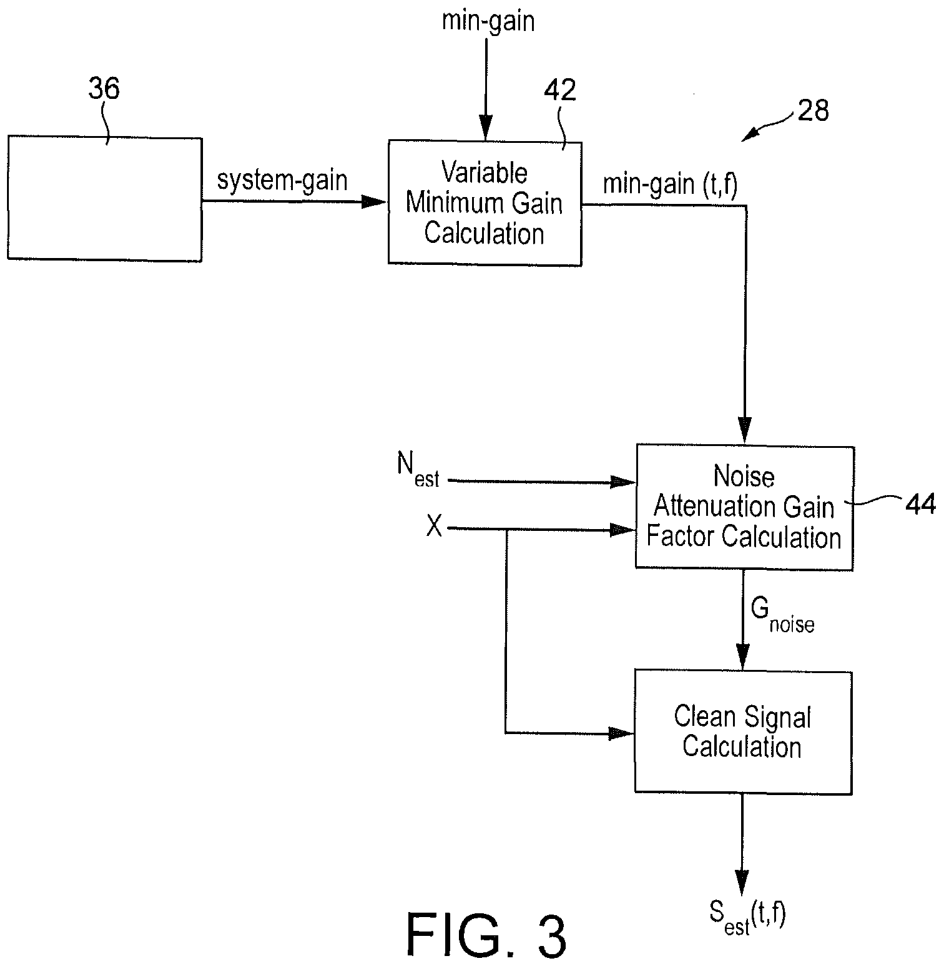
min-gain
36
42
28
system-gain
Variable Minimum Gain Calculation
min-gain (t,f)
N est
X
Noise Attenuation Gain Factor Calculation
44
G noise
Clean Signal Calculation
S est(t,f)

FIG. 3

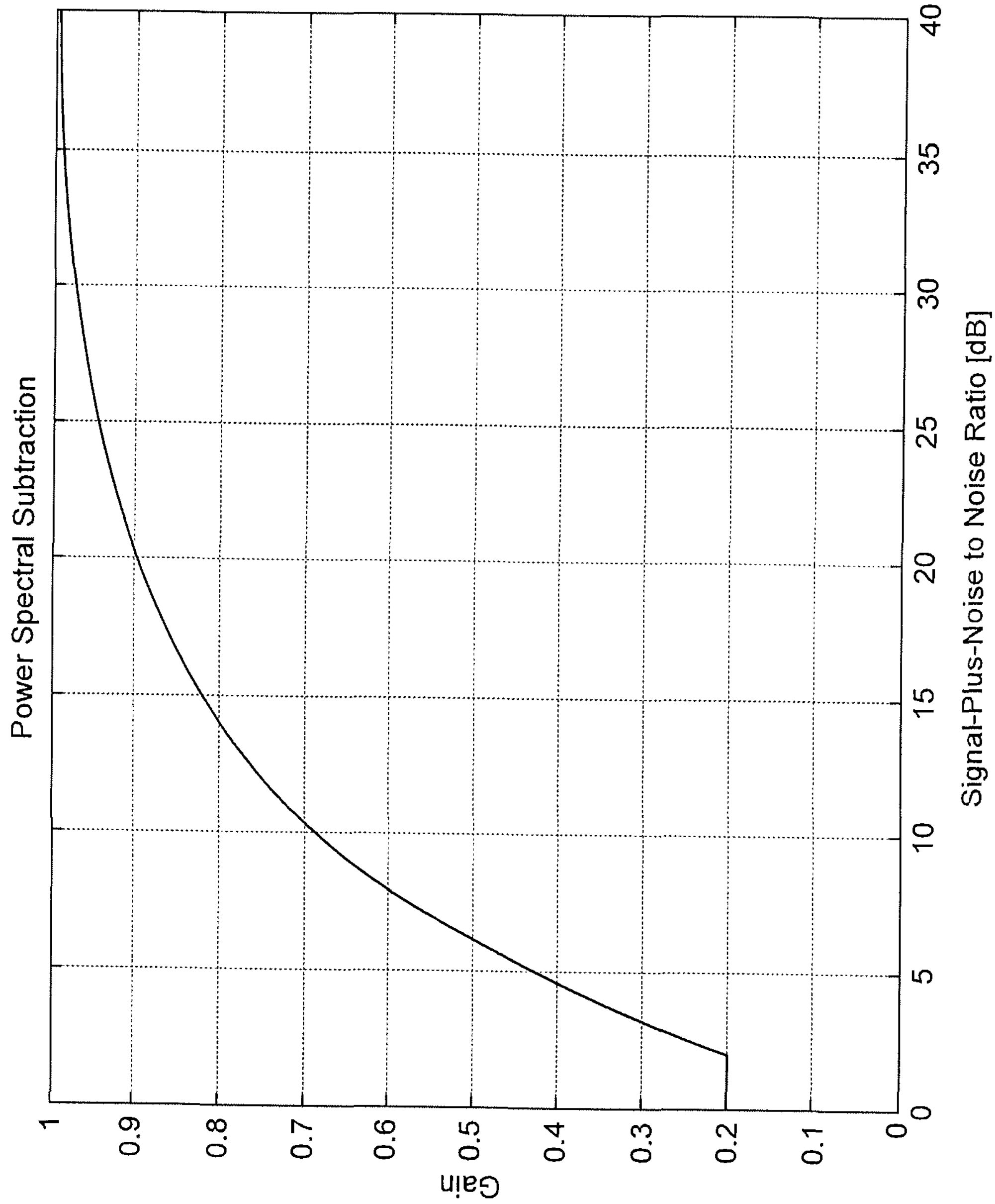
Power Spectral Subtraction
Gain
1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
5
10
15
20
25
30
35
40
Signal-Plus-Noise to Noise Ratio [dB]

FIG. 4

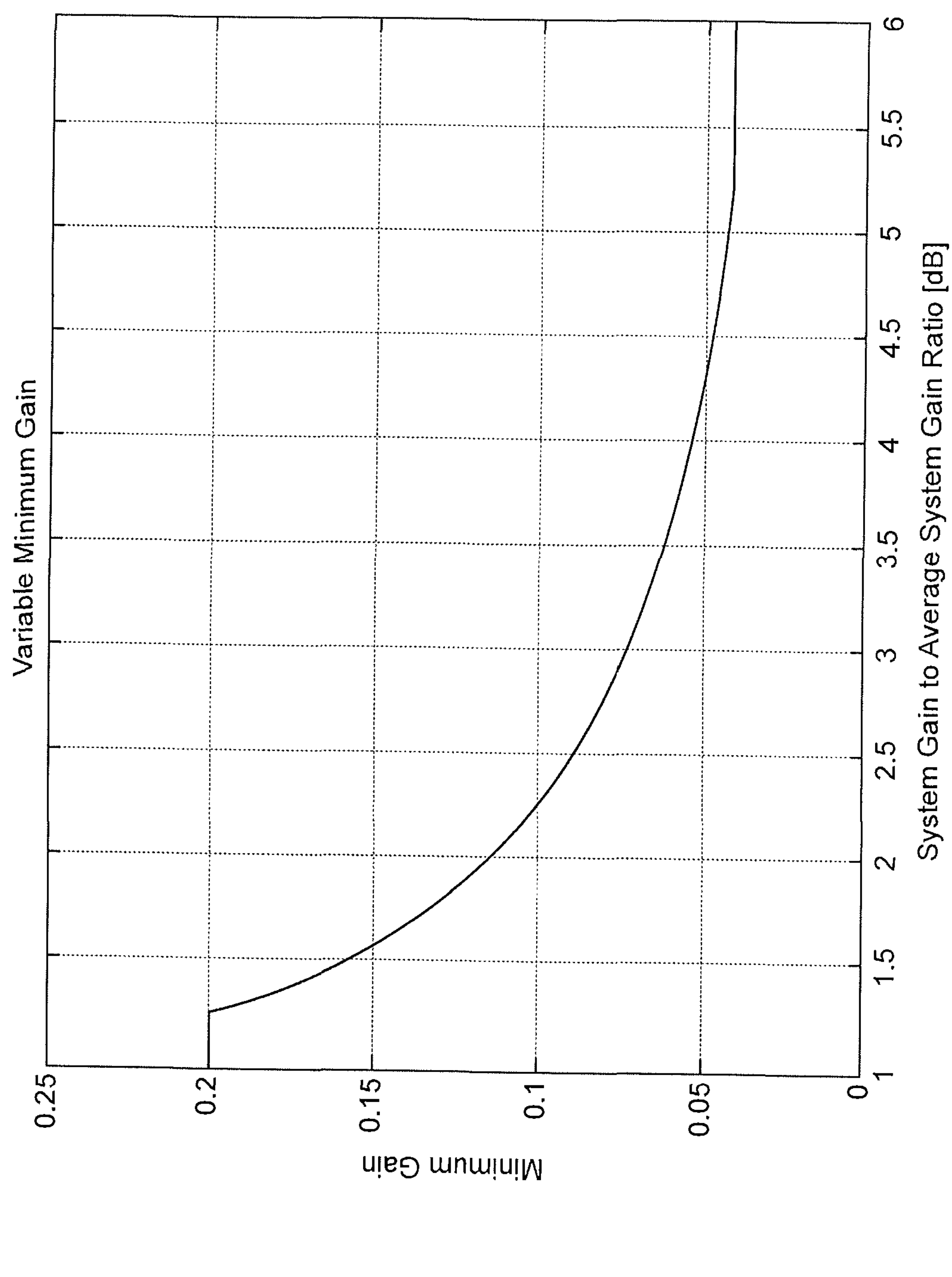
Variable Minimum Gain
Minimum Gain
0.25
0.2
0.15
0.1
0.05
0
1
1.5
2
2.5
3
3.5
4
4.5
5
5.5
6
System Gain to Average System Gain Ratio [dB]

FIG. 5

PROCESSING AUDIO SIGNALS

FIELD

The invention relates to processing audio signals, particularly but not exclusively in the case of a communication session between a near end device and a far end device.

BACKGROUND

Communication systems allow users to communicate with each other over a network. The network may be, for example, the Internet or public switched telephone network (PSTN). Audio signals can be transmitted between nodes of the network, to thereby allow users to transmit and receive audio data (such as speech data) to each other in a communication session over the communication system.

A user device may have audio input means such as a microphone that can be used to receive audio signals such as speech from a user. The user may enter into a communication session with another user, such as a private call (with just two users in the call) or a conference call (with more than two users in the call). The user's speech is received at the microphone, processed and is then transmitted over a network to the other users in the call.

As well as the audio signals from the user, the microphone may also receive other audio signals, such as background noise, which are unwanted and which may disturb the audio signals received from the user.

The user device may also have audio output means such as speakers for outputting audio signals to near end user that are received over the network from a far end user during a call. Such speakers can also be used to output audio signals from other applications which are executed at the user device, and which can be picked up by the microphone as unwanted audio signals which would disturb the speech signals from the near end user.

In addition, there might be other sources of unwanted noise in a room, such as cooling fans, air conditioning systems, music playing in the background and keyboard taps. All such noises can contribute to disturbance to the audio signal received at the microphone from the near end user for transmission in the call to a far end user.

In order to improve the quality of the signal, such as for use in the call, it is desirable to suppress unwanted audio signals (the background noise and the unwanted audio signals output from the user device) that are received at the audio input means of the user device. Various noise reduction techniques are known for this purpose including, for example, spectral subtraction (for example, as described in the paper “Suppression of acoustic noise in speech using spectral subtraction” by S. F. Bool IEEE Trans. Acoustics, Speech, Signal Processing (1979), 27(2):, pages 113-120.

Another difficulty that can arise in an acoustic system is “howling”. Howling is an unwanted effect which arises from acoustic feedback in the system. It can be caused by a number of factors and arises when system gain is high.

It is an aim of the present invention to reduce howling without unnecessarily interfering with optimisation of the perceptual quality of noise reduction techniques used in audio signal processing.

<u>SUMMARY OF THE INVENTION</u>

According to one aspect of the present invention there is provided a method of reducing noise in a signal received at a processing stage of an acoustic system, the method comprising, at the processing stage:

identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system;

providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

applying the noise attenuation factor to a component of the signal at that frequency.

In the described embodiment, the step of identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system is carried out by estimating a respective system gain of the acoustic system for each of a plurality of frequencies in the received signal. This allows one or more frequencies which cause the higher system gain to be identified. In this case, it is not necessary to actually calculate an average system gain – it will be apparent that the highest system gains are above the average.

Alternatively, the frequency can be identified based on known characteristics of a device including the processing stage. For example, it might be apparent that a particular component of the device (for example, a loudspeaker) has a problematic resonant frequency which would cause howling.

Alternatively, rather than estimating a system gain, the system gain can actually be measured. For example, it could be estimated or measured based on the echo path. References to "system gain" herein encompass an estimated system gain and/or a measured system gain.

Although it is possible to obtain advantages from the invention by attenuating only one frequency which is likely to predispose the acoustic system to howling, it is particularly advantageous if a respective system gain of the acoustic system is calculated for each of a plurality of frequencies in the received signal, and a noise attenuation factor is provided for each of the plurality of frequencies. In that case, each noise attenuation factor can be applied to a respective component of the signal at that frequency. In this way, the system gain spectrum of the acoustic system can be taken into account.

In the described embodiment, each of the plurality of frequencies lies in a frequency band, and the system gain and noise attenuation factor for each frequency is applied over the whole of the frequency band containing that frequency. In a practical embodiment frequencies in the range 0 to 8 KHz are handled over 64 or 32 bands of equal width.

The invention is particularly useful where the signal received at the processing stage is speech from a user. In that case, the speech is processed in time intervals, for example, frames, and the respective system gain and noise attenuation factors are provided for each of the plurality of frequencies in each frame.

The system gain can be estimated by multiplying all gains that are applied in the system, including the gain in the echo path which can be either an estimated or predetermined.

In a described embodiment, the noise attenuation factor which is provided for each frequency is selected as the maximum of a first and second noise attenuation factor. In that case, the first noise attenuation factor can be calculated based on a signal-plus-noise to noise ratio of the signal, and the second noise attenuation factor can be a variable minimum gain factor based on the system gain. In that embodiment of the invention, the effects of the invention are only felt at signal components with lower signal-plus-noise to noise ratios where the variable minimum gain factors are provided as the noise attenuation factors for the different frequencies. For components with higher signal-plus-noise to noise ratios, the noise attenuation factor is calculated and provided in a way which causes the noise reduction to gently reduce as the signal-plus-noise to noise ratio increases, thus leaving behind near end speech without any significant reduction or equalisation.

The variable minimum gain factor can be based on the system gain according to a function which selects a minimum of a ratio of maximum system gain to average system gain and at least one predetermined value. The function can be multiplied by a constant minimum gain factor.

The noise reduction method discussed herein can be applied on a signal for playout that has been received from the far end in a communication network, or be applied partly on the far end signal and partly on a signal received at the near end (for example, by an audio input means at a user device).

The invention also provides in another aspect, an acoustic system comprising:

audio input means arranged to receive a signal;

a signal processing stage connected to receive the signal from the audio input means; the signal processing stage comprising:

means for identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system;

means for providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

means for applying the noise attenuation factor to a component of the signal at that frequency.

A further aspect provides a signal processing stage for processing an audio signal, the signal processing stage comprising:

means for identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system;

means for providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

means for applying the noise attenuation factor to a component of the signal at that frequency.

Another aspect provides a user device comprising audio input means for receiving an audio signal from a user;

a signal processing stage for processing the signal; and

a wireless communication means for transmitting the processed signal from the user device to a remote device, the signal processing stage as defined above.

According to another aspect of the present invention, there is provided a method of reducing noise in a signal received at a processing stage of an acoustic system, the method comprising, at the processing stage:

estimating or measuring a respective system gain of the acoustic system for at least one frequency in the received signal;

providing a noise attenuation factor for reducing noise in the signal at that frequency, the noise attenuation factor being based on the system gain measured or estimated for that frequency; and

applying the noise attenuation factor to a component of the signal at that frequency.

Preferably, the system gain is estimated or measured for each of a plurality of frequencies in the received signal, and a respective noise attenuation factor is provided and applied for respective components of the signal at each frequency, the noise attenuation factor for each frequency being based on the system gain estimated or measured for that frequency.

In the following embodiments of the invention, there is achieved the advantage of system gain reduction arising from equalisation by noise attenuation, while adapting to the actual conditions. This means that any acoustic effect on the system gain spectrum from the room is taken into account.

For a better understanding of the present invention and to show how the same may be carried into effect, reference will now be made by way of example to the accompanying drawings.

Description of the Drawings

Figure 1 is a schematic diagram of a communication system;
Figure 2 is a block diagram of a user device;
Figure 3 is a schematic function diagram of a noise attenuation technique;
Figure 4 is a graph of gain vs. signal plus noise to noise ratio; and
Figure 5 is a graph of minimum gain vs. system gain to average system gain ratio.

Description of the Preferred Embodiments

In the following described embodiments of the invention, a technique is described wherein a continuously updated estimate of the system gain spectrum is applied to adapt a noise reduction method to apply more noise suppression in parts of the spectrum where the system gain is high. By applying greater noise suppression in parts of the spectrum where the system gain is high, the system gain over those parts is reduced and thus robustness to howling is increased. Before describing the particular embodiments of the present invention, a context in which the invention can usefully be applied will now be described with reference to Figure 1, which illustrates a communication system 100.

A first user of the communication system (User A 102) operates a user device 104. The user device 104 may be, for example a mobile phone, a television, a personal digital assistant ("PDA"), a personal computer ("PC") (including, for example, Windows™, Mac OS™ and Linux™ PCs), a gaming device or other embedded device able to communicate over the communication system 100.

The user device 104 comprises a central processing unit (CPU) 108 which may be configured to execute an application such as a communication client for communicating over the communication system 100. The application allows the user device 104 to engage in calls and other communication sessions (e.g. instant messaging communication sessions) over the communication system 100. The user device 104 can communicate over the communication system 100 via a network 106, which may be, for example, the Internet or the Public Switched Telephone Network (PSTN). The user device 104 can transmit data to, and receive data from, the network 106 over the link 110.

Figure 1 also shows a remote node with which the user device 104 can communicate over the communication system 100. In the example shown in Figure 1, the remote node is a second user device 114 which is usable by a second user 112 and which comprises a CPU 116 which can execute an application (e.g. a communication client) in order to communicate over the communication network 106 in the same way that the user device 104 communicates over the communications network 106 in the communication system 100. The user device 114 may be, for example a mobile phone, a television, a personal digital assistant ("PDA"), a personal computer ("PC") (including, for example, Windows™, Mac OS™ and Linux™ PCs), a gaming device or other embedded device able to communicate over the communication system 100. The user device 114 can transmit data to, and receive data from, the network 106 over the link 118. Therefore User A 102 and User B 112 can communicate with each other over the communications network 106.

Figure 2 illustrates the user device 104 at the near end speaker in more detail. In particular, Figure 2 illustrates a microphone 20 receiving a speech signal from user 22. The microphone can be a single microphone or a microphone array comprising a plurality of microphones and optionally including a beamformer. As is known, a beamformer receives audio signals from the microphones in a microphone array and processes them in an attempt to improve the signal in a wanted direction in comparison to signals perceived to be coming from unwanted directions. This involves applying a higher gain in a desired direction.

Signals from the microphone (whether with or without a beamformer) are applied to a signal processing stage 24. The signal processing stage 24 includes a plurality of signal processing blocks, each of which can be implemented in hardware or software or a combination thereof as is deemed appropriate. The blocks can include, for example, a digital gain block 26, a noise attenuation block 28 and an echo canceller block 30.

A loud speaker 32 is provided to provide audio signals 34 intended for the user 102. Such signals can come from a far end speaker to be output to a user, or can alternatively come from the user device itself as discussed earlier. In a situation where signals output by the loudspeaker 34 come from a far end user such as user 112, they can be processed before being emitted by the loudspeaker by signal processing circuitry and for the sake of convenience the loudspeaker is shown connected to signal processing circuitry 24 in Figure 2. Optionally, they can be processed using the noise attenuation technique described below.

After signal processing, the signals input by the user 102 and picked up by the microphone 20 are transmitted for communicating with the far end user 112.

The signal processing circuitry 24 further includes a system gain estimation block 36. As discussed in more detail later, and as distinct from known system gain estimation blocks, block 36 estimates system gain taking into account the shape of the system gain spectrum. That is, the system gain varies with frequency. Estimates of system gain for different frequencies are supplied to the noise attenuation block 28.

Howling is a symptom of having feedback with a system gain higher than 1 somewhere in the frequency spectrum. By reducing the system gain at this frequency, the howling will stop. Very often, a resonating frequency in the loudspeaker, microphone or echo path will be much larger than average and will be what is limiting the robustness to howling. The system gain is estimated by taking into consideration the blocks involved in system processing (including for example the digital gain block, echo canceller, and background noise attenuation block), and in particular, uses information from the echo path estimated in the echo canceller attenuation block which provides information about the room in which the device is located. The shape of the spectrum is usually dominated by the estimated echo path, as the transfer function of the echo path includes the transfer function of the loudspeaker where resonating frequencies often occur. In Figure 2, the estimated echo path is denoted by arrow 40.

By estimating system gain spectrum contribution from the near end side, it is possible to obtain knowledge about which parts of the spectrum are more likely to dominate in generation of a howling effect. When two similar devices 104, 114 are being used in a call, this half-side information can be very accurate in terms of knowing which part of the spectrum will be dominating as the resonating frequencies will coincide on the two devices.

The estimate of system gain spectrum supplied to the noise attenuation block 28 is used to modify operation of the noise attenuation method, as discussed below.

Signal processing is performed on a per frame basis. Frames can, for example, be between 5 and 20 milliseconds in length and for the purpose of noise suppression be divided into spectral bins, for example, between 64 and 256 bins per frame. Each bin contains information about a signal component at a certain frequency, or in a certain frequency band. For dealing with wideband signals, the frequency range from 0 to 8 kHz is processed, divided into 64 or 32 frequency bands of equal width. It is not necessary that the bands are of equal width – they could for example be adjusted to better reflect the critical bands of the human hearing such as done by the Bark scale.

Ideally, for speech, each frame is processed in real time and each frame receives an updated estimate of system gain for each frequency bin from system gain block 36. Thus each bin is processed using an estimate of system gain specific to that frame and the frequency of that bin.

Figure 3 illustrates according to one example, how a noise attenuation gain factor can be calculated to take into account frequency based estimates of system gain.

It will be appreciated that Figure 3 illustrates various functional blocks which can be implemented in software as appropriate. A variable minimal gain calculation block 42 generates a variable minimum gain value min_gain(t,f)) at time t and frequency f. The variable minimum gain value is generated based on the system gain system_gain and a fixed minimum gain value min_gain as in equation 1.

min_gain(t,f) = min_gain * f(system_gain(t,f))

In the variable minimum calculation block the function, f(·) , of the system gain according to one example is as given in equation 2.

f(system_gain(t,f)) = min(max(system_gain(t,f) / avg_system_gain(t), 1.25, 5,25) – 0.25)$^{-1}$.

This function has the effect of lowering the variable minimum gain value min_gain(t,f) when the system gain is high in the current frequency band. As will be clear from the following, this has the effect of more noise attenuation in the bands with the highest local system gain.

The variable minimum gain value is supplied to a noise attenuation gain factor calculation block 44. This block calculates a noise attenuation gain factor $G_{noise}(t,f)$ at time t and frequency f. The gain factor G_{noise} takes into account a noise level estimate N_{est} and the signal received from the microphone X, representing the signal plus noise incoming from the microphone.

A first noise attenuation gain factor is calculated according to equation 3.

$$G_{noise}(t,f) = ((X(t,f)^2 - N_{est}(t,f)^2) / X(t,f)^2) = (1 - (X(t,f)^2 / N_{est}(t,f)^2)^{-1}).$$

In classical noise reduction, such as for example, power spectral subtraction as in the example above, the coefficient $S_{est}(t,f)$ at time t and frequency f of the estimated clean signal is calculated as the square root of the noise attenuation gain multiplied with the squared coefficients of the signal plus noise – that is, as in equation 4 where equation 3 provides the noise attenuation gain factor G_{noise}.

$$S_{est}(t,f) = \mathrm{sqrt}(G_{noise}(t,f) * X(t,f)^2)$$

Thus, $S_{est}(t,f)$ represents the coefficient of the best estimate of a clean signal for transmission to the far end after signal processing.

The noise attenuation gain factor G_{noise} can be lower limited for improving perceptual quality as in equation 5.

$$G_{noise}(t,f) = \max(1 - (X(t,f)^2 / N_{est}(t,f)^2)^{-1}, min\_gain(t,f)).$$

That is, the noise attenuation gain factor calculated according to equation 3, is only applied to the extent that it is above a minimum gain value min_gain (f,t)..

In existing noise reduction techniques, the minimum gain value is fixed at min_gain, and could take, for example, a constant value of approximately .2. In contrast, embodiments of the present invention vary the minimum gain value as has been described to provide an individual minimum gain for each frequency band, such that the minimum gain value can be lowered when the local system gain for that band is high. The minimum gain value is a function of the system gain spectrum which is adapted over time, such that it tracks any changes that may occur in the system gain spectrum.

By incorporating spectral system gain equalisation in the noise reduction method, it is provided that in a state of no speech activity, the left-behind noise is equalised by applying more noise reduction in frequency bands where the system gain is high and thereby reducing the system gain in those bands. This is shown in equation 5, which indicates that the noise attenuation gain factor G_{noise} is the maximum of the variable minimum gain value and the value calculated using the signal-plus-noise to noise ratio. This has the effect of allowing a higher noise reduction (lower G_{noise}) when the signal-plus-noise to noise ratio is low. When the signal-plus-noise to noise ratio is high, however, for example in the case of near end activity, the effect of the variable minimum gain factor is overtaken by the conventional calculation of the noise attenuation factor G_{noise}, which reduces the noise attenuation as the signal to noise ratio increases. In such a case, near end speech is thus left without any significant reduction or equalisation.

Figure 4 illustrates the case where the minimum gain is a constant value of approximately .2 and shows the effect on the gain factor G_{noise} as the signal plus noise to noise ratio increases. As G_{noise} approaches 1, the noise attenuation decreases until it is virtually zero as the signal plus noise to noise ratio increases.

Figure 5 is graph showing how the minimum gain varies as a function of the system gain according to equation 2.

CLAIMS:

1. A method of reducing noise in a signal received at a processing stage of an acoustic system, the method comprising, at the processing stage:

identifying at least one frequency at which a system gain of the acoustic system is above an average system gain of the acoustic system;

providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

applying the noise attenuation factor to a component of the signal at that frequency.

2. A method according to claim 1, wherein the step of identifying said at least one frequency comprises estimating a respective system gain of the acoustic system for each of a plurality of frequencies in the received signal.

3. A method according to claim 2, wherein a respective noise attenuation factor is provided for each of the plurality of frequencies, and each noise attenuation factor is applied to a respective component of the signal at that frequency.

4. A method according to claim 3, wherein each of the plurality of frequencies lies in a frequency band, the system gain and noise attenuation factor for each frequency being applied over the frequency band containing the frequency.

5. A method according to any of claims 2 to 4, wherein the system gain is estimated from an echo path in the acoustic system.

6. A method according to any preceding claim, wherein the step of identifying at least one frequency is based on known characteristics of a device which includes the processing stage.

7. A method according to claim 1, wherein the step of identifying said at least one frequency comprises measuring a system gain.

8. A method according to any preceding claim, wherein the steps of identifying, providing and applying are carried out at repeated time intervals.

9. A method according to claim 8, wherein the repeated time intervals are frames of the received signal.

10. A method according to any of claims 2 to 4, wherein the step of estimating a respective system gain comprise estimating the gain in each of a plurality of processing blocks in the signal processing stage.

11. A method according to any preceding claim, wherein the step of providing a respective noise attenuation factor comprises calculating a first noise attenuation factor based on a signal (or signal-plus-noise) to noise ratio of the received signal at the at least one frequency, calculating a second noise attenuation factor based on the system gain for that frequency, and;

providing the one of the first and second noise attenuation factors with the higher value.

12. A method according to any preceding claim, wherein the noise attenuation factor is based on the system gain according to a function of the system gain which comprises selecting a minimum of a ratio of maximum system gain to average system gain and at least one predetermined value.

13. A method according to claim 12, wherein the noise attenuation factor is based on the system gain by a multiple of said function and a constant minimum gain value.

14. A method according to claim 8 or 9, wherein each time interval comprises a plurality of bins, each bin containing components of the received signal in a particular frequency band, wherein a noise attenuation factor is determined for each bin.

15. A method according to any preceding claim, when applied to a signal which has been received from a remote device.

16. A method according to any preceding claim, wherein the method is applied to a signal input at a device including the processing stage.

17. An acoustic system comprising:

audio input means arranged to receive a signal;

a signal processing stage connected to receive the signal from the audio input means; the signal processing stage comprising:

means for identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system;

means for providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

means for applying the noise attenuation factor to a component of the signal at that frequency.

18. An acoustic system according to claim 17 wherein the means for identifying at least one frequency comprises means for estimating a respective system gain of the acoustic system for each of a plurality of frequencies in the received signal.

19. An acoustic system according to claim 18, wherein a respective noise attenuation factor is provided for each of the plurality of frequencies, and is applied to a respective component of the signal at that frequency.

20. An acoustic system according to claim 18 or 19, which comprises an echo path, wherein the means for estimating the system gain bases the estimate of system gain on the echo path.

21. An acoustic system according to any of claims 17 to 20, comprising a microphone as said audio input means.

22. An acoustic system according to any of claims 17 to 21, comprising a loud speaker for providing audio signals to a user.

23. A signal processing stage for processing an audio signal, the signal processing stage comprising:

means for identifying at least one frequency which causes a system gain of the acoustic system to be above an average system gain of the acoustic system;

means for providing a noise attenuation factor for reducing noise in the signal for the at least one frequency, the noise attenuation factor for the at least one frequency based on the system gain for that frequency; and

means for applying the noise attenuation factor to a component of the signal at that frequency.

24. A signal processing stage according to claim 23, wherein the means for identifying at least one frequency comprises means for estimating a respective system gain of the acoustic system for each of a plurality of frequencies in the received signal.

25. A signal processing stage according to claim 23, wherein a respective noise attenuation factor is provided for each of the plurality of frequencies, and is applied to a respective component of the signal at that frequency.

26. A signal processing stage according to claim 23, which comprises an echo path, wherein the means for estimating the system gain bases the estimate of system gain on the echo path.

27. A user device comprising:

audio input means for receiving an audio signal from a user;

a signal processing stage for processing the signal; and

a wireless communication means for transmitting the processed signal from the user device to a remote device, the signal processing stage being in accordance with any of claims 23 to 26.

INTELLECTUAL
PROPERTY OFFICE

Application No:	GB1102704.2	**Examiner:**	Mrs Hannah Sylvester
Claims searched:	1-27	**Date of search:**	9 August 2012

Patents Act 1977: Search Report under Section 17

Documents considered to be relevant:

Category	Relevant to claims	Identity of document and passage or figure of particular relevance
A	-	US2010/151787 A1 (MOTOROLA SOLUTIONS INC [US])
A	-	US2005/226444 A1 (PEAVEY ELECTRONICS CORP [US])
A	-	US2005/207567 A1 (FORGENT NETWORKS INC [US])
A	-	GB2293078 A (YAMAHA CORP [JP])
A	-	EP2337376 A1 (YAMAHA CORP [JP])
A	-	US5406635 A (NOKIA MOBILE PHONES LTD [FI])

Categories:

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
&	Member of the same patent family	E	Patent document published on or after, but with priority date earlier than, the filing date of this application.

Field of Search:

Search of GB, EP, WO & US patent documents classified in the following areas of the UKC^X :

Worldwide search of patent documents classified in the following areas of the IPC

G10L; H04M; H04R

The following online and other databases have been used in the preparation of this search report

WPI EPODOC TXTE

INTELLECTUAL
PROPERTY OFFICE

International Classification:

Subclass	Subgroup	Valid From
None		