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(54) **Title**: ANTI-FEEDBACK DEVICE

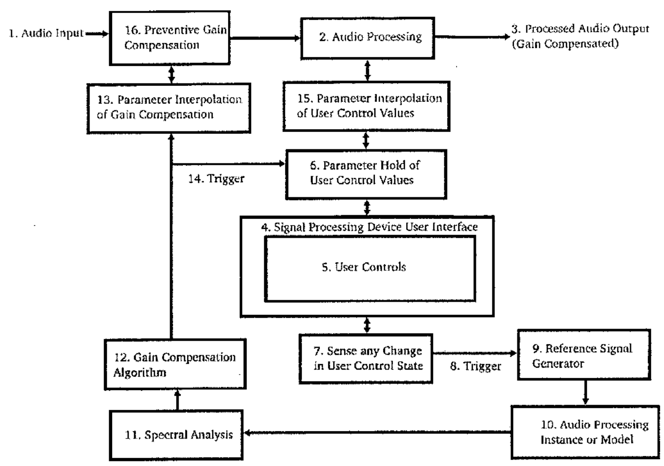

Fig. 4

(57) **Abstract**: A feedback-prevention method for a system with a signal processor comprising the steps of inputting at least one reference signal (9); calculating a spectral model (11) of the system according to the or each reference signal; generating a preventative gain compensation factor (12) according to the spectral model of the system; applying the preventative gain compensation factor (16) to the input or output of the signal processor (2); and outputting a gain-compensated audio output (3).

WO 2009/138754 A1

Anti-Feedback Device

The present invention relates to a method and apparatus for preventing feedback in audio systems.

Feedback is the result of returning the output signal of a system to its input. When this output signal is equal or greater than the input signal a "howling" artefact is produced. In an acoustic system these artefacts are introduced due to the feedback path and can be positive and negative feedback contributions. A simplified diagram of an acoustic feedback system is shown in Figure 1. The source signal is picked up by a microphone, transformed by equalization, amplified and played back through a speaker at the output. This is then attenuated and delayed as the output is transmitted through air, and summed with the input signal. $H_{ETOT}(x)$ is the electronic feed-forward transfer function of the system, and it is the result of the product of the individual transfer functions of the signal chain given by the microphone equalizer amplifier and speaker. $H_{ATOT}(x)$ is the acoustic transfer function of the system.

Undesired feedback phenomena, known as howling or the Larsen effect, is an established acoustic problem (see S. H. Antman, "Extension to the theory of howlback in reverberant rooms." Acoustical Society of America, vol. 2.14, 2.7; 5.13, p. 399, 1965.). Howling is a state in which system gain exponentially increases out of control, causing an undesired audible pitch. The feedback causes the audio system to behave in an unstable manner and hence should be avoided at all costs.

In practical situations, howling occurs when the sound engineer changes the levels on a mixing desk resulting in the gain for a particular frequency range going above a critical level. To stop this, the sound engineer will adjust the global gain of the audio system, adjusting linearly all frequency levels down to an acceptable level. To achieve maximum gain before feedback, audio operators have relied mainly on equalizers, delay and feedback cancellation techniques. Cancellation techniques are applied after the processed audio signal is output. The problem is that the onset of the feedback is exponential so the sound engineer has to respond quickly. This is a particular problem with live performances.

In recent years, understanding of the acoustic feedback phenomena and when feedback can be improved has advanced substantially. There are several known methods for eliminating feedback, as discussed below.

A common method of achieving maximum gain before feedback is by the use of feedback cancellation. Currently there are four main feedback-controlling techniques (whitepaper, D. Troxel, "Understanding Acoustic Feedback & Suppressors," Rane Corporation 2005). The first one consists of frequency shifting the output signal so that the electronic transfer function is out of alignment with the acoustic transfer function, this causes a destructive interaction between the input and the acoustic feedback path, which effectively reduces feedback. In practice it can achieve up to 3 dBs increase in gain before feedback. This method is effective for speech applications but is not suitable for music. This is because it modifies pitch, which would result in undesired atonal music.

The second feedback control technique is the all-pass filter approach. This is used to invert the phase of a potential feedback frequency. Unfortunately, this technique is only useful with low delay systems with a prominent resonance. When applied to a system with flat frequency response it causes the feedback to jump endlessly from one section of the spectrum to other. For this reason its use is very limited.

Also known is the adaptive filter modelling (for example: S. Kamerling, et al, "A New Way of Acoustic Feedback Suppression," in *104th Audio Engineering Society Convention*, 1998). This uses technology based on echo-cancellation, aimed at telecommunication applications. The main idea is to subtract the far-end speech from the near-end speech. When the model is accurate it can achieve up to 10dBs of added gain before feedback. Due to the closed loop nature of the acoustic audio system the residual error of this process is highly correlated to the signals involved, and this can cause noticeable artefacts. When the model deviates it can introduce distortion and artefacts. It can even cause undesired feedback artefacts, which should not have been there. For this reason it has mainly been applied for speech systems where conditions are controlled. It is currently not considered a good candidate for sound reinforcement.

Finally, there is the adaptive notch filter method (for example: whitepaper, D. Troxel, "Understanding Acoustic Feedback & Suppressors" Rane Corporation 2005), which consists of a series of fixed and non-fixed notch filters, which filter out feedback frequencies when detected. The system performance is a trade-off between speed of detection and accuracy, and can notch out program material if a feedback discrimination system is not implemented properly or the system is overused. This method is highly effective and is widely used on sound reinforcement applications. Unfortunately, it does not offer any extra gain before feedback for a flat frequency response system.

Currently, there is no optimal feedback cancellation method for music, which offers a substantial improvement in gain before feedback without the detrimental side effects discussed above.

The present invention provides an improved anti-feedback device and method for a full range of audio systems, which alleviates the problems described above.

In one aspect the invention provides an anti-feedback method for a system with a signal processor comprising the steps of inputting at least reference signal; calculating a spectral model of the system according to the or each reference signal; generating a preventative gain compensation factor according to the spectral model of the system; applying the preventative gain compensation factor to the input or output of the signal processor; and outputting a gain-compensated output.

The present invention allows a user to achieve maximum gain before feedback, whilst allowing a user to interact with the system to achieve an output, which is proportional to the input from the user control, without being concerned about introducing unwanted feedback artifacts. The method of the present invention is a technique for real-time magnitude gain normalization of a changing linear system, which preserves system stability and eliminates the production of feedback artefacts, rather than cancelling out the feedback. The method prevents undesirable feedback artefacts being created rather than suppressing feedback artefacts after they have occurred.

Preferably, the preventative gain compensation factor is calculated according to a normalization constant, wherein the normalization constant is determined to be the inverse of the maximum of the transfer function of an output generated in response to the reference signal.

Preferably, the feedback-prevention method of the present invention further comprises the steps of inputting an input signal into a memory block, and triggering release of the input signal from the memory block to the signal processor in response to generation of the preventative gain compensation factor.

Preferably, the reference signal is triggered by a change in at least one user input parameter.

The triggering of the method of the present invention when a change occurring at the user interface is detected ensures correct normalization every time the linear state of the system has changed.

Preferably, the reference signal is a measurement signal such as an impulse, sweep, step signal or noise.

Preferably, the feedback-prevention method is carried out either at clock speed or at sample rate speed.

Preferably, the feedback-prevention method of the present further comprises the step of displaying the gain of the preventative gain-compensated output.

Preferably, the feedback-prevention method of the present invention generates the preventative gain compensation factor according to a spectral analysis of the or each reference signal.

Preferably, the signal processor is an audio processor.

In one embodiment, the present invention provides a computer-readable carrier medium carrying computer readable instructions for performing the feedback-prevention method according to any preceding claim.

The feedback-prevention method of the present invention can be implemented in software, in hardware as a firmware, or as a solid-state device.

In a second aspect, the present invention provides a feedback-prevention device for a system with a signal processor comprising an input for inputting at least one reference signal; a calculating means for calculating a spectral model of the system according to the or each input and the or each corresponding reference signal; a generating means for generating a preventative gain compensation factor according to the spectral model of the system; a compensating means for applying the preventative gain compensation factor to the input or the output of the signal processor; and an output means for outputting a gain-compensated output.

Preferably, the feedback-prevention device further comprises a user interface having one or more control means.

More preferably, the user interface is detachable.

An advantage of the user interface detachment is that the method can be implemented on analogue systems by interfacing the analogue user interface with analogue to digital converters and by transferring the results to the audio device using digital to analogue converters.

Preferably, the feedback-prevention device further comprises a sensing means for sensing a change in one of the control means.

Preferably, the feedback-prevention device further comprises storage means for storing at least one data signal input into the device.

Preferably, the feedback-prevention device further comprises a display means for displaying the gain of the preventative gain-compensated audio output.

Preferably, the signal processor of the feedback-prevention device is an audio processor

An embodiment of the present invention is described with reference to the figures, in which:

Figure 1 illustrates a known model of a sound reinforcement feedback system; and

Figure 2 is a flow diagram illustrating the proposed normalization technique using a truncated impulse response according to the present invention; and

Figure 3 is an amplitude/frequency plot showing example results of the transfer functions of an un-normalized and a normalized response according to the present invention; and

Figure 4 is a flow diagram illustrating the feedback prevention method of the present invention.

Referring to the acoustic model in Figure 1, the system will introduce undesired howling artifacts if equation 1 is satisfied.

$H_{ETOT}(x) \cdot H_{ATOT}(x) >= 1$ (1)

If, for example, the equalizer transfer function gain, $H_{eEQ}(x)$, is 0dBs for all x, i.e. the function is flat and the overall electronic transfer function of the system $H_{ETOT}(x)$ is on the marginal condition before howling, increasing the gain of a range of frequencies, Δx_1, will introduce an undesired feedback artefact.

To stop feedback either: 1) the gain of the same range of frequencies, Δx_1, is reduced so that $H_{eEQ}(x)$ is again 0dBs for all x or 2) all frequencies are reduced linearly so that just Δx_1 is 0dBs, and all other frequencies are below 0dBs. The second of these is a normalization technique, which enables the relative gain changes to be preserved while eliminating feedback by forcing the transfer function of a linear system to have a maximum peak of 0dBs. This will preserve the stability of the system.

The method is useful in situations where the maximum gain before feedback is needed. The system normalization is independent of the input signal. The present invention eliminates feedback entirely by automating this process so that the global gain is reduced before the gain of a frequency range goes above the critical level. The method automatically calculates the appropriate gain that should be applied in order to maintain maximum unitary gain. The method uses an impulse measurement to generate a mathematical model of the system and calculate a normalizing factor. The normalizing factor is then applied prior to the signal leaving the audio system.

The method of the present invention is based on an algorithm that automatically adjusts the transfer function of a system, or element in a system, to ensure that the maximum gain in the transfer function is below 0dBs, where 0dbs is a threshold established by the user. For the purpose of not producing feedback artefacts, the threshold must be below the maximum gain before the feedback limit. This means that for a stable system, which has no feedback artefacts occurring, the method can be used to ensure the system remains stable even when modifying the transfer function, due to the transfer function being automatically normalised to its previous, stable, optimal value.

The method relies on determining the maximum value in the transfer function every time there are changes to a system. This is done by obtaining the truncated impulse response measurement of a model of the system to be normalised and multiplying the whole transfer function of the system to be normalised by a factor equal to: 1/ "value of the maximum".

The normalization technique could be applied to: audio systems; improving close loop robot control while keeping stability; improving close loop motor control; improving and correcting measurement devices while ensuring no distortion and improved loop control; for loudness control; data acquisition and instrumentation improvement while ensuring no introduction of distortion or instability problems; improving reverberation algorithms without rendering them unstable; improving radar and sonar response and stability; for novice audio operators while ensuring stability; for altering the frequency characteristics of a IIR filter without causing the system to be become unstable or begin oscillating; for filter design in general; avoiding distortion (for example in a pre-amplifier); equalizing RF loops while avoiding increases in gain; re-equalizing Sigma Delta Modulators without causing them to become unstable; improving the loop characteristics of analogue to digital converters; altering the characteristics of phase lock loops without making them go unstable; telecommunications, conference rooms, recording studios, radio stations; and normalization of full audio mixing board devices.

The system uses an unsolved difference equation usually derived, but not limited to, a *Z* domain mathematical model. An approximation derived from laboratory measurements can also be used as a target measurement system. Therefore, the model is generated according to the system being normalised and is accurately determined to maximise efficiency. For a less efficient implementation of the present invention, a duplicate of the system being normalised will suffice. An approximation, which will compromise resolution, can also be used where applicable. When using this normalisation technique for multiple channels, the use of a linear phase system design is recommended to avoid channel phase interaction.

Referring to the known acoustic model of Figure 1, a system will introduce undesired howling artifacts if equation 2 is satisfied.

$$H_{ETOT}(x)\cdot H_{ATOT}(x)>=1 \qquad (1)$$

Referring to the method of the present invention, as shown in Figure 4, measurement is performed by inputting an impulse to the mathematical model and obtaining its maximum through measurement on its output.

It is known from Fourier theory and linear system theory that:

$i(t)= FFT^{-1}(H(w))$ (2)

where $i(t)$ is the output impulse response of the system, $FFT^{-1}(t)$ is the inverse Fourier transform and $H(w)$ is the transfer function of the system under study. By applying the following identity, where $f(t)$ represents an arbitrary time domain function,

$f(t)= FFT^{-1}(FFT(f(t)))$ (3)

and given that the input $x(t)=\delta(t)$ where $\delta(t)$ is an impulse then we can say that $y(t)=i(t)$, therefore:

$H(w)= FFT(y(t))$ (4)

Thus, the transfer function of a complex system whose input is an impulse response is given by performing the FFT of the output.

In other words the normalization constant can be found by applying an impulse to a mathematical model of the audio system, such as a Z domain function. Then a simple FFT is applied to the output. The resulting output can now be searched for the maximum value. In practice, only searching half the FFT data is necessary. The inverse of the obtained value is the normalization constant to be applied to the input.

The algorithm for implementing the automatic maximum gain normalization technique is presented in Figure 2. In a standard system, the user interface would be connected directly to the audio processing device. For demonstrating the algorithm, we have detached the user interface and stored the corresponding coefficients coming from the interface in a memory block called the "fade in" parameters block. This memory block sends the coefficients to the audio processing device once the normalization constant has been found. The coefficients together with the normalization constant are transferred using a linear interpolation algorithm that ensures a soft, modulation-free transition to the next system state. The method of the present invention can be implemented on analogue systems by interfacing the analogue user interface with analogue to digital converters and by transferring the results to the audio device using digital to analogue converters.

The algorithm sends an impulse to initiate the feedback prevention method every time a change in the user interface has been detected. This ensures a correct normalization every time the linear system state has changed. Thus, it is possible to calculate correctly the normalization value even if the transfer function order changes, for example when bypassing certain sections of an equalizer or even if the system design has changed, such as changing a filter in real time from a peak/notch to a shelf filter.

An example of the results given by the feedback prevention method is shown in Figure 3, the dash-dotted (-·-·-) line represents the 0dB threshold for maximum gain before feedback, the dashed line (- - -) represents the transfer function of a non-normalized acoustic system and the full line is the transfer function after applying the feedback prevention method of the present invention.

The system uses a side processing architecture, contained outside the audible audio path, which measures the impulse response and gradually interpolates the obtained normalising factor with the old normalisation value to achieve real time normalisation with no audio artefacts. Gradual interpolation is performed simultaneously between the previous user interface control values and the new user interface control values; this interpolation is synchronised with the normalisation factor interpolation. The approach is to create a model of the audio device so that every time the user interface of the audio system is altered, by the user, during normal operation, through a button, knob, fader or other control means, an impulse response measurement is calculated from the revised model. This impulse measurement is used to derive the factor which is then be used to normalise the incoming signal before it is goes out of the audio system (or output signal, both are equivalent).

The model applied according to the present invention will be different for each type of audio device, but can be produced relatively simple, as it can be derived from the equations which represents the settings on the audio device or even from a matched audio device or model (a copy of the device which shares the same user interface and the same transfer function characteristics). The method can be applied to normalise any linear system, which has a transfer function that is being continuously changed by the user. For example the normalization technique has applications where not going beyond a normal value is required for example on speech, or music, in live, broadcast and recording for example in radio station or recording studios where going pass the 0dB limits causes distortion on the transmission/recording medium, or in live situations where adding gain to the electronic transfer function of the system can introduce feedback artefacts.

A truncated impulse response measurement is produced from a matched transfer function (matched to the transfer function of the audio device to be normalized). The user interface is linked to this match mathematical model which is used to calculating the normalizing factor and once it is calculated the user interface values together with the normalisation value are "faded in" into the audio device. This process eliminates any audible artefacts before the impulse is passed to the audio device, while maintaining the frequencies ratios of the transfer function modified by the user

The method of the present invention can be implemented either at clock speed or at sample rate speed. The only section of the algorithm that needs to be revised if the linear system is changed is the memory sector containing the mathematical model of the audio device. This gives the automatic maximum gain normalization technique the capability of being implemented as a solid-state chip, which can be interconnected to memory containing the model.

Given the flexibility of the algorithm and the universality of the normalizing approach, the system can be implemented in software, in hardware, as a firmware or as a solid-state device with a changeable memory sector containing the mathematical model of the system to be normalised. It has potential for being part of standard channel normalisation on mixing consoles or as a stand-alone product. Normalized equalisation is one of the most practical uses for the device.

It is envisaged that the method and device of the present invention can also be applied to detect feedback by detecting when the gain of a system is above a predetermined threshold. For example, the present invention can be used to detect when sounds levels are above a permitted threshold.

Referring to Figure 4, the audio processing device of the present invention consists of an audio input, an audio processing section and a signal output. The output is the result of processing the input audio signal. The audio processing unit further comprises a signal-processing interface, which consist of one or more user controls. However, it is envisaged that in some embodiments the unit will not include a user-interface. The user controls, contained within the user interface, are capable of modifying parameters inside the audio processing algorithm. Every time the user modifies the state of the user control parameters this affects the overall stability of the system and causes undesired artifacts, such as retro-alimentation of a signal, also known as "howlback". The present invention is a self-compensating device and method, which maintains system stability, thus preventing undesired phenomena such as howlback.

The audio processing device comprises one or more audio inputs 1, an audio processor 2, and a signal output 3, which outputs a processed signal output. The processed signal output is a gain compensated signal. The output provided at 3 is tuned to prevent any undesired feedback phenomena. The audio processing device has a signal processing user interface 4, which is controlled by a one or more user controls 5. The processing device compensates for any changes to the audio input/s 1 made via the audio controls 5. The method carried out by the device of the present invention is carried out every time the user changes the state of any of the user controls 5.

If a user changes a parameter of the audio input using the user controls 5, the altered parameter is inputted to a "parameter hold" buffer 6. Simultaneously, a sensor 7 detects any change in the user control state. A change in the user controls 5 is sensed and a trigger signal 8 is sent to a reference signal generator 9. The reference signal can be any measurement signal (not necessarily an audio signal) such as a sweep; noise; impulse or any other reference signal, which have spectrally meaningful, known characteristics. All parameters of the reference signal are known. For example, in the case of a digital impulse, the reference signal would be a single sample with a known gain of, for example, 1. The reference signal generally runs at rates faster than audio speed and is inputted into a model 10.

The model 10 is capable of processing the reference signal at a higher data rate then the audio processor 2 of the device. The audio processing model 10 duplicates the "black box" containing the algorithm of the audio processor 2. It is not necessary to know the full details of the algorithm of the audio processor 2 because the method of the present invention can compensate for feedback by using a copy of the processor's algorithm. That is, if the audio processing algorithm of the processor 2 is unknown, a duplication or reconstruction of the algorithm is used. If the algorithm is known, then a more compact mathematical model approximation can be used. That is, the model 10 can process the algorithm, for example the known algorithm of an equalizer, to simplify the algorithm and optimize the performance of the system.

Figure 4 shows a model 10 wherein a copy of the original signal processing algorithm of the audio processing device 2 is made. The model 10 is capable of working at audio signal rate, but is also capable of running at a higher data rate; the reconstruction can run at speeds higher than audio rate and improves the flexibility and efficiency of the method. By way of example, the model 10 receives a reference signal 00000001000000, where 1 corresponds to the impulse. If the output of the audio processing device 2 is 10000000000000 then the system is characterized by a delay of size 0000000. The example is given in the time domain, but it is possible to express the signals in the frequency domain. If the present method is applied to an equalizer, for example, then the output of the model 10 comprises the spectral shape or frequencies to be applied by the equalizer according to instructions input by a user.

The spectral analysis 11 comprises a time to spectral domain transform such as a Fast Fourier transform (FFT), a constant Q-transform, or any spectral transform capable of obtaining the amplitude spectrum of the output of the model 10. The spectral amplitude of the output is then passed to a gain compensation algorithm 12 and the inverse of the maximum magnitude of the spectral amplitude is determined. This value is used to calculate the required gain compensation factor that must be applied to the signal parameter to prevent feedback artifacts. For example, if the amplitude of the reference impulse input to the model 10 is 1 and the maximum amplitude found following the time to spectral domain transform is x2 then the gain compensation factor is ½. As previously discussed with reference to equation 1, when the maximum spectral peak is less than or equal to 1, the overall gain of the device will be compensated to prevent undesired howling artifacts. The transfer function is normalized to have a maximum peak of 1 whilst conserving the ratios between frequencies.

The gain compensation factor 12 is then sent to a signal parameter gain compensation interpolator 13, which determines a preventive gain compensation factor 16 according to the audio input 1. The signal passed to the audio processor 2 comprises the audio input 2 of the signal processing device and the preventative gain compensation factor 16. The audio output 3 from the audio processor is thus compensated in order to prevent feedback loop instability.

It is envisaged that the preventative gain compensation factor 16 can be applied to the audio input 1 or to the audio output 2. Both methods will provide an equivalent result.

Simultaneously, the output of a gain compensation factor 12 creates a trigger 14 to send a release order to the parameter holding buffer 6. The parameter holding buffer 6 reacts to the trigger 14 to output the user control parameters, stored therein in the first step of the above-described method. The audio parameters are sent to the user control parameter interpolator 15, which interpolates between a previous user control state and the new user control state, as requested via the user controls 5. The signal parameter gain compensation interpolator 13, referred to above, interpolates between a previous gain compensation factor and a new gain compensation factor. This achieves a processed audio output signal 3 which is proportional to the input from the user control settings 5, whilst ensuring system stability and preventing audible feedback artifacts.

The device of the present invention allows the user to change the user controls without introducing undesired feedback loop artifacts. This means that, for example in the case of a signal processor such as an equalizer, the user can boost the signal bands gain as high as desired without introducing feedback due to incorrect use of the device's controls.

The method described above can also be applied to hearing aids, where a change of equalization in the hearing aid can cause a feedback loop between the device's microphone and the headphone transducer. The application of the above-described feedback prevention method is used to stop the creation of feedback loop artifacts.

The gain compensation method and device of the present invention can also be used in sound recording applications, where changing the user parameters via mixing apparatus would normally incur the risk of exceeding the dynamic range of the recording device. The application of the above-described method ensures that the audio output signal will not surpass the maximum dynamic range of the recording system.

In live music applications, the application of the above-described method ensures that an audio engineer can be confident that, regardless of the changes made to the settings of the device, the preventative gain compensation factor of the present invention ensures that the stability of the live audio output is maintained.

The device and method of the present invention can also be applied to automatic mixing, because the device can ensure stability at all times even when a machine is attempting to autonomously generate changes to a mixture, and so the audio input from the user parameters. The device can also be used to maintain the stability of a feedback loop of a closed loop electronic circuit, such as an amplifier circuit.

The normalization gain compensation technique described above can be used to compensate for any processing device in which there is a feedback loop involved and is not limited to audio devices. The system can be used to maintain the overall gain of the system under changing signal input conditions. The above described embodiments have been given by way of example only, and the skilled reader will naturally appreciate that many variations could be made thereto without departing from the scope of the present invention, as defined by the claims.

Laboratory testing

Laboratory testing was carried out on a real acoustic system and provided successful results using the above-described method. A self-powered studio monitor playing wideband-recorded music was used as a source. The speaker was placed 10cm away from an omni-directional, flat frequency response microphone. Care was taken to keep the source level set such that microphone diaphragm distortions were avoided. The microphone was then connected to a soundcard interfaced to the anti-feedback device of the present invention.

The output of the system was connected to a line driver to control the overall amplification gain of the system. A self-powered studio monitor was used as the main sound reinforcement speaker. Care was also taken to avoid electronic and acoustic distortion of the system. With the equalizer remaining "flat" the system was driven to the marginal state of maximum gain before feedback. Numerous boosts and cuts were applied to the equalizer and compensation of up to -50dBs were achieved without howling. It was also observed that only a 3dB margin was required for avoiding howlback due to artifacts introduced by high quality factor (Q) on the low frequency range.

The testing, which was implemented on a six biquadratic parametric filter implementation, has shown the suitability of the method and device of the present invention for use in sound reinforcement applications.

Claims

1. A feedback-prevention method for a system with a signal processor comprising the steps of:

   inputting at least one reference signal;

   calculating a spectral model of the system according to the or each reference signal;

   generating a preventative gain compensation factor according to the spectral model of the system;

   applying the preventative gain compensation factor to the input or output of the signal processor;

   outputting a gain-compensated output.

2. A feedback-prevention method according to claim 1 wherein the preventative gain compensation factor is calculated according to the maximum gain in the transfer function such that the maximum gain is maintained below a predetermined threshold.

3. A feedback-prevention method according to claim 1 or claim 2 wherein the preventative gain compensation factor is calculated according to a normalization constant, wherein the normalization constant is determined to be the inverse of the maximum of the transfer function of an output generated in response to the reference signal.

4. A feedback-prevention method according to any preceding claim further comprising;

   inputting an input signal into a memory block, and triggering release of the input signal from the memory block to the signal processor in response to generation of the preventative gain compensation factor.

5. A feedback-prevention method according to any preceding claim wherein the reference signal is triggered by a change in at least one user input parameter.

6. A feedback-prevention method according to claim 4 wherein the reference signal is a measurement signal such as an impulse, sweep, step signal or noise.

7. A feedback-prevention method according to any preceding claim wherein the method is carried out either at clock speed or at sample rate speed.

8. A feedback-prevention method according to any preceding claim further comprising the step of displaying the gain of the preventative gain-compensated output.

9. A feedback-prevention method according to any preceding claim wherein the preventative gain compensation factor is generated according to a spectral analysis of the or each reference signal.

10. A feedback-prevention method according to any preceding claim wherein the signal processor is an audio processor.

11. A computer-readable carrier medium carrying computer readable instructions for performing the feedback-prevention method according to any preceding claim.

12. A feedback-prevention device for a system with a signal processor comprising:

an input for inputting at least one reference signal;

a calculating means for calculating a spectral model of the system according to the or each input and the or each corresponding reference signal;

a generating means for generating a preventative gain compensation factor according to the spectral model of the system;

a compensating means for applying the preventative gain compensation factor to the input or the output of the signal processor;

an output means for outputting a gain-compensated output.

13. A feedback-prevention device according to claim 12 further comprising a user interface having one or more control means.

14. A feedback-prevention device according to claim 13 further comprising a sensing means for sensing a change in at least one of the control means.

15. A feedback-prevention device according to any of claims 12 to 14 further comprising storage means for storing at least one data signal input into the device.

16. A feedback-prevention device according to any of claims 12 to 15 further comprising a display means for displaying the gain of the preventative gain-compensated audio output.

17. A feedback-prevention device according to any of claims 12 to 16 wherein the signal processor is an audio processor

18. A feedback-prevention method substantially as hereinbefore described or referred to in Figures 2 and 4.

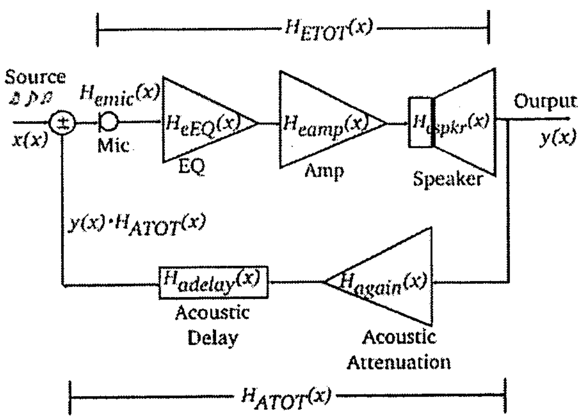

Fig. 1

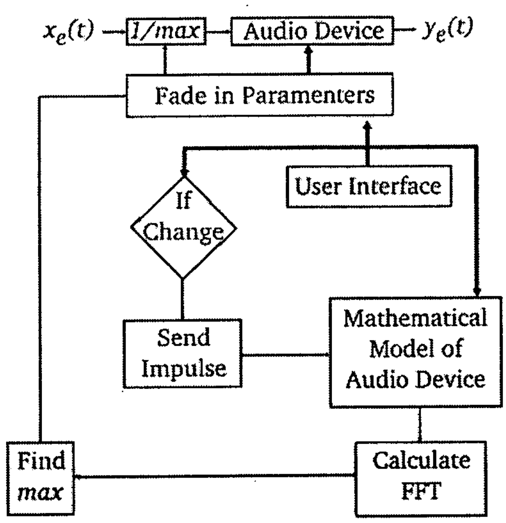

Fig. 2

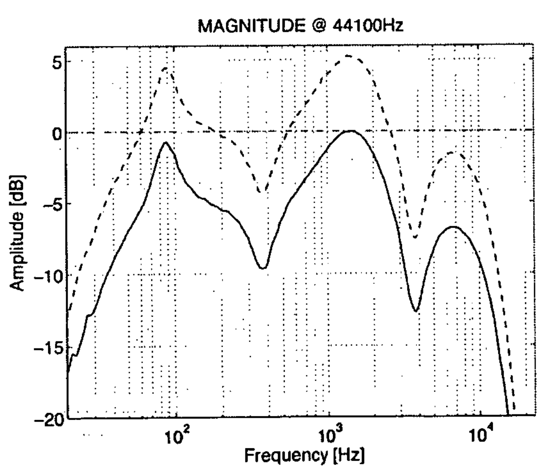

Fig. 3

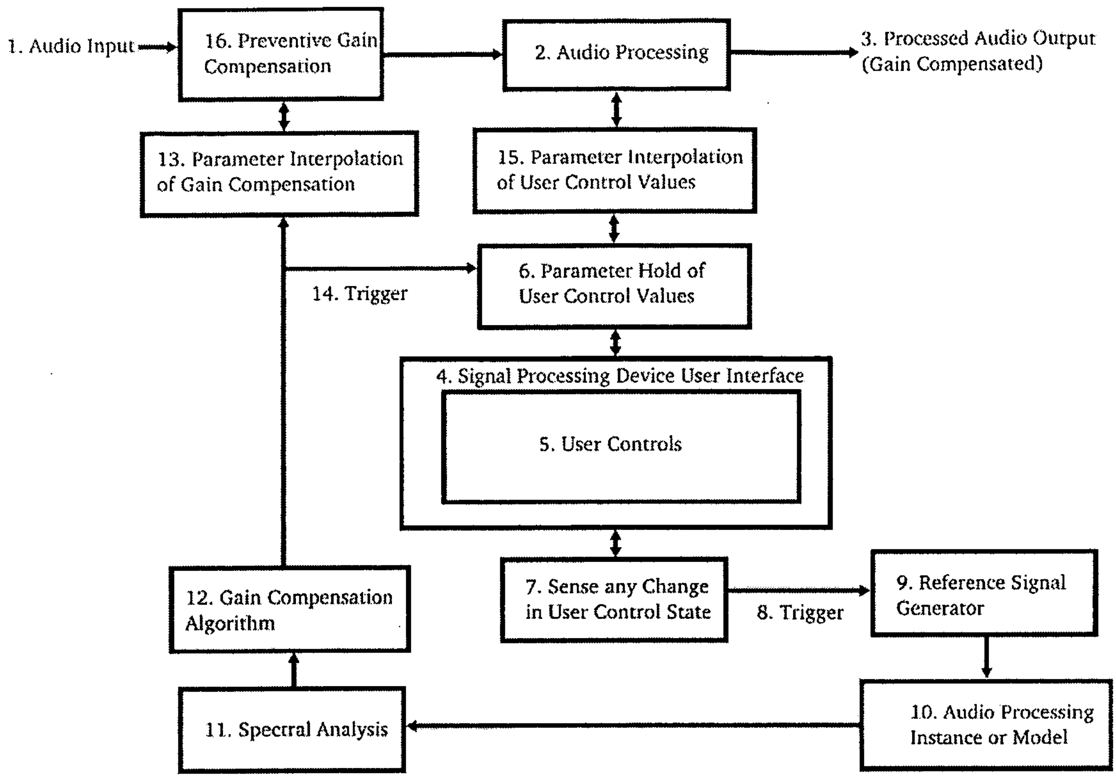
1. Audio Input
16. Preventive Gain Compensation
2. Audio Processing
3. Processed Audio Output (Gain Compensated)
13. Parameter Interpolation of Gain Compensation
15. Parameter Interpolation of User Control Values
14. Trigger
6. Parameter Hold of User Control Values
4. Signal Processing Device User Interface
5. User Controls
12. Gain Compensation Algorithm
7. Sense any Change in User Control State
8. Trigger
9. Reference Signal Generator
11. Spectral Analysis
10. Audio Processing Instance or Model

Fig. 4

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2009/001218

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04R3/02

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04R

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

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, IBM-TDB, INSPEC, COMPENDEX

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 98/51066 A (ERICSSON GE MOBILE INC [US]) 12 November 1998 (1998-11-12) page 3, line 2 - page 5, line 20 page 14, line 21 - page 16, line 23 figures 1,2	1,2, 11-18
A		3-10

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

[X] See patent family annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

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"O" document referring to an oral disclosure, use, exhibition or other means

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

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
25 August 2009	02/09/2009

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Fax: (+31-70) 340-3016	Meiser, Jürgen

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2009/001218

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	VICEN-BUENO R ET AL: "Acoustic feedback reduction based on Filtered-X LMS and Normalized Filtered-X LMS algorithms in digital hearing aids based on WOLA filterbank" INTELLIGENT SIGNAL PROCESSING, 2007. WISP 2007. IEEE INTERNATIONAL SYMPOSIUM ON, IEEE, PISCATAWAY, NJ, USA, 3 October 2007 (2007-10-03), pages 1-6, XP031228517 ISBN: 978-1-4244-0829-0 page 1, right-hand column, last paragraph - page 3, right-hand column, last paragraph figures 1,2	1,2, 11-18
A		3-10

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
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