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(54) **Audio processing device and method of performing frequency characteristic correction processing for an audio input signal**

(57) An audio processing device comprises an input terminal (2) configured to receive an audio input signal from an audio source, an equalizing circuit (7) having an input and an output, the input coupled to the input terminal (2) and configured to execute frequency characteristic correction processing for a signal received at the input, and an output circuit (8, 9, 3) coupled to the output of the equalizing circuit (7) and configured to supply an audio output signal to a speaker for audio output from the speaker. A signal analyzing circuit (11, 12) is coupled to receive the audio input signal and configured to analyze a frequency characteristic of the audio input signal for detecting the presence of a signal power of the audio input signal at at least one frequency (F1, F2) of a room mode, wherein the signal analyzing circuit (11, 12) is coupled to a control terminal of the equalizing circuit (7) for dynamically adapting the frequency characteristic correction processing at the at least one frequency of a room mode over time when the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode is detected. The device is capable of handling steady state compensation for room modes without affecting short lasting low frequency audio signals.

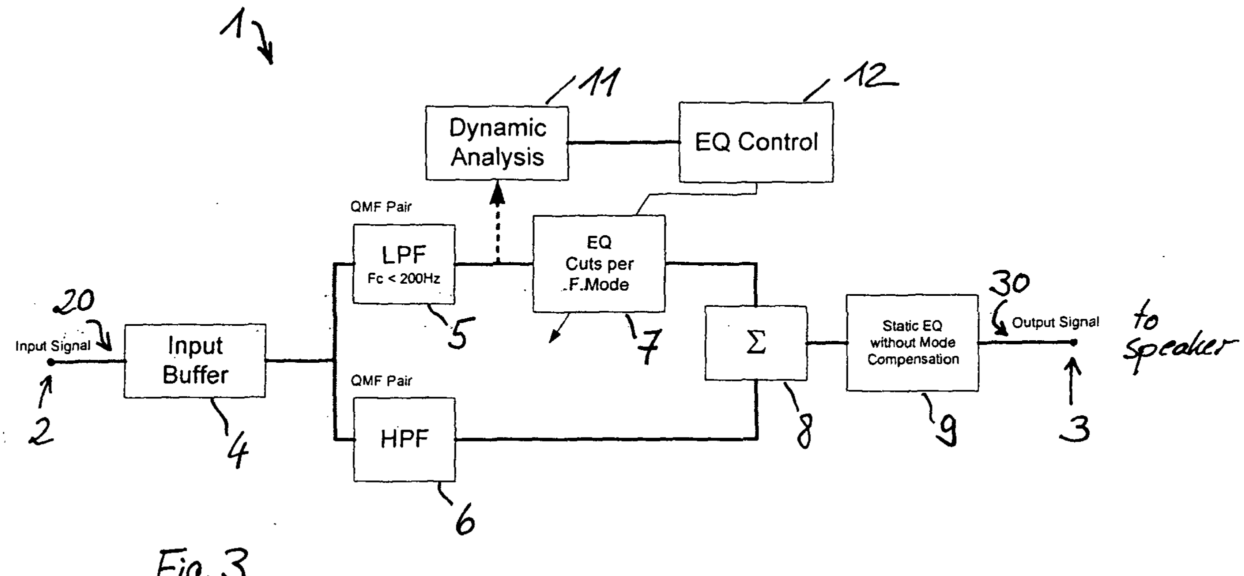

Fig. 3

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Description

Background of the Invention

Field of the Invention

[0001] The present invention is directed to an audio processing device configured to receive an audio input signal from an audio source, and comprising an equalizing circuit configured to execute frequency characteristic correction processing for a received audio signal and an output circuit coupled to an output of the equalizing circuit and configured to supply an audio output signal to a speaker for audio output from the speaker. Further, the invention is directed to a method of performing frequency characteristic correction processing for an audio input signal.

Description of the Related Art

[0002] In audio systems, typically an electrical signal from an audio source having audio information content (hereinafter referred to as an "audio signal") is provided to one or multiple speakers which transform the received audio signal into sound. Given an audio system installed in a room environment, the sound is reflected by reflecting objects such as a wall of the room, wherein sound is reflected back into the room and interferes with sound directly emitted from the speakers at the position of the listener, which may cause coloration of the sound heard by the listener. In order to adapt the audio system to the specific room environment, an audio system may use an equalizing circuit configured to execute frequency characteristic correction processing for a received audio signal. Such equalizing circuit is typically used for filtering and/or correcting an overall frequency response of the audio signal having specific signal frequencies in order to exert influence on the sound spectrum at the position of the listener, for example, to compensate for reflection effects as mentioned above.

[0003] Generally, room parameters of a room environment which influence sound reflection are not known prior to installation of the audio system since such room parameters vary depending on a particular environment in which the audio system may be installed. Varying room parameters may be, for example, a result of varying distances of opposed walls of different room environments in which the audio system may be installed, or different absorbing/reflecting objects present in the room. In this respect, it is known in the art to measure such frequency response between speaker and listener position in the particular audio system installation and to set a respective filter parameter of the equalizing circuit accordingly. An output circuit is coupled to an output of the equalizing circuit and supplies an audio output signal having a correspondingly corrected frequency characteristic to the speakers for audio output from the speakers.

[0004] In the above mentioned context concerning reflection of emitted sound, so-called room modes are of particular relevance. Room modes can be identified by peaks and dips in the frequency response of the acoustic transfer function between speaker and listening position, typically only at low frequencies (e.g., below 150 Hz) where their density is not too high over space. Room modes are caused by the formation of standing waves inside a room or cavity, like in a vehicle or car cabin. Standing waves will appear in cases where the wavelength of the sound wave is on the order of the room dimension (half wavelength relationship) and also reflections, e.g. at room walls, are present.

[0005] An example of a particular room mode for a given cubic room or cavity is shown in Fig. 1. The room R comprises a distance D between the opposed walls W which is a multiple of $\lambda/2$ of the wavelength λ of the room mode RM. A room mode may significantly cause sound coloration at the listener's position since room modes may cause amplifying or attenuation effects for signal components at the particular room mode frequency.

[0006] However, room modes typically do not occur at a steady level, but vary in their intensity over time due to the interaction of excitation energy, reflections and absorptions in the room. In this context, Fig. 2 shows an exemplary room mode measurement over time, taken from "Comparison of Modal Equalizer Design Methods", Poju Antsalo1, et. al., AES Convention 2003 March 22-25, page 5.

Fig. 2 shows that all room modes decay over time within a particular so-called decay time, which is in the present example approximately 500ms. It is also possible to see that several modes are present and they decay also differently in the room, thus each having a different decay time. The highest peaks determine the dominating modes in the room. A similar observation can be made for the so-called onset time of a room mode which indicates the time for forming the corresponding room mode upon emitting a particular sound signal which forms the corresponding room mode. Such onset time of a room mode is typically in the order of the above mentioned decay time for that room mode.

[0007] The above mentioned equalizing circuit may be used, for example, to filter or correct signal components of the audio signal having frequencies where room modes occur in order to compensate for the amplifying effect of the corresponding room mode at the listener's position. For example, signal components of the audio signal may be filtered and/or corrected so as to gain a flattened or "equalized" signal response over the frequency band without peaks or dips. However, such frequency characteristic correction processing or equalizing processing (herein referred to as equalizing compensation or EQ compensation in the following) may negatively affect the sound at low frequencies as a matter of steady state EQ compensation of room modes. Sound like short drum or short bass sounds are affected by the EQ compensation in the same manner as longer lasting low frequency sound, such as long (e.g. lasting 800 ms or longer) and deep sound produced by a pipe organ, which excites and creates a corresponding room mode. Therefore, conventional EQ compensation is a compromise between the optimization for steady state (long lasting sound) and the dynamic case (short lasting sound).

[0008] In JP 2000-261900 A an acoustic device using a steady state sound field correction method is provided. Particularly, microphones are installed at optional positions in a room, multiple speakers measure acoustic characteristics at the positions, analyze the measured values and are installed at prescribed positions in the room. A system control circuit controls audio signals supplied to the speakers on the basis of the analyzed value, and controls a standing wave at an optional listening position so as to correct the frequency characteristic thereby making the frequency characteristic flat at the listening position through changing the frequency characteristic in the room. Furthermore, the sound pressure level at the measured position can closely be made constant thereby reducing a sound pressure difference within a listening plane.

Summary of the Invention

[0009] It is an object of the present invention to provide an audio processing device and a method of performing frequency characteristic correction processing for an audio input signal which are capable of improving the sound quality for different types of low frequency audio signals which may excite room modes.

[0010] The invention is directed to an audio processing device according to the features of claim 1. Further, the invention is directed to a method of performing frequency characteristic correction processing for an audio input signal according to the features of claim 11. Embodiments and advantageous features of the invention are mentioned in the dependent claims.

[0011] Particularly, embodiments of an audio processing device according to the invention comprise an input terminal configured to receive an audio input signal from an audio source, an equalizing circuit having an input and an output, wherein the input is coupled to the input terminal and the equalizing circuit is configured to execute frequency characteristic correction processing for a signal received at the input, and an output circuit coupled to the output of the equalizing circuit and configured to supply an audio output signal to a speaker for audio output from the speaker. A signal analyzing circuit is coupled to receive the audio input signal and is configured to analyze a frequency characteristic of the audio input signal for detecting the presence of a signal power of the audio input signal at at least one frequency of a room mode. The signal analyzing circuit is further coupled to a control terminal of the equalizing circuit for dynamically adapting the frequency characteristic correction processing at the at least one frequency of a room mode over time when the presence of the signal power of the audio input signal at the at least one frequency of a room mode is detected.

[0012] Further, embodiments of the invention include a method of performing frequency characteristic correction processing for an audio input signal, comprising the steps of analyzing a frequency characteristic of an audio input signal for detecting a presence of a signal power of the audio input signal at at least one frequency of a room mode, performing, in an equalizing process, frequency characteristic correction processing for the audio input signal, dynamically activating and deactivating the frequency characteristic correction processing at the at least one frequency of a room mode over time if the presence of the signal power of the audio input signal at the at least one frequency of a room mode has been detected, and supplying from the frequency characteristic correction processing an audio output signal to a speaker for audio output from the speaker.

[0013] With adapting of the EQ compensation over time for a specific frequency of a room mode, it is possible to account for the dynamic influences monitored with respect to the formation of room modes. The equalizing processing may be dynamically adapted depending on the type of sound. Sound like short drum or short bass sound may not be affected or may be less affected by the EQ compensation, whereas longer lasting low frequency sound is subjected to EQ compensation to reduce the negative effects of room modes. Thus, the invention accounts for the effects that short lasting low frequency sound, in contrast to long lasting low frequency sound, will not excite or create room modes so that, in this case, the corresponding signal components of the audio signal having these frequencies will not be subjected to EQ compensation which, if applied, may distort the sound in the absence of a corresponding room mode.

[0014] In an embodiment of the invention, the signal analyzing circuit stores a predetermined onset time for at least one frequency of a room mode. If the signal analyzing circuit detects the presence of the signal power of the audio input signal at the at least one frequency of a room mode, the signal analyzing circuit determines a duration of the detection of the presence of the signal power and compares the duration with the stored onset time of the corresponding room mode. If the duration is equal to or greater than a predetermined time threshold, the signal analyzing circuit controls the equalizing circuit for activating the frequency characteristic correction processing for the at least one frequency of a room mode.

[0015] According to another embodiment, the signal analyzing circuit further determines a signal power of the audio input signal at the at least one frequency of a room mode compares the determined signal power with a signal power threshold value, wherein if the determined signal power is equal to or greater than the signal power threshold value, the signal analyzing circuit controls the equalizing circuit for activating the frequency characteristic correction processing at the at least one frequency of a room mode. In this way, it is accounted for the effect that sound which exceeds a particular signal power will more likely excite a room mode than sound which is lower than the signal power threshold.

[0016] According to another embodiment, the signal analyzing circuit stores a predetermined decay time for at least one frequency of a room mode. The signal analyzing circuit further determines a signal power of the audio input signal at the at least one frequency of a room mode and compares the determined signal power with a signal power threshold value. If the signal analyzing circuit has previously controlled the equalizing circuit for activating the frequency characteristic correction processing at the at least one frequency of a room mode, the signal analyzing circuit determines whether the determined signal power is lower than the signal power threshold value. If the determined signal power is lower than the signal power threshold value, the signal analyzing circuit controls the equalizing circuit for deactivating the frequency characteristic correction processing for the at least one frequency of a room mode after the decay time has elapsed counted from the point of time of when the signal analyzing circuit has determined that the determined signal power is lower than the signal power threshold value.

[0017] In an embodiment of the invention, the signal analyzing circuit stores a table containing multiple frequencies of respective room modes, wherein the table associates a respective predetermined onset time and a respective predetermined decay time of the room modes with a corresponding one of the multiple frequencies.

[0018] The audio processing device may further include a buffer circuit coupled between the input terminal and the equalizing circuit for successively storing a number of samples of the audio input signal, wherein the signal analyzing circuit is coupled to the buffer circuit for analyzing a frequency characteristic of the audio input signal. For example, the buffer circuit has a length which is capable to store samples of the audio input signal during a period of time which is in the order of a room mode onset or decay time.

[0019] Particularly, the audio processing device of the invention also includes a low pass filter coupled between the input terminal and the equalizing circuit and having an output providing an output signal to the equalizing circuit containing only lower frequencies of the audio input signal. A high pass filter may be coupled between the input terminal and the output circuit and in parallel to the series connection of the low pass filter and equalizing circuit, wherein the output circuit comprises a summation circuit for adding a signal provided from the high pass filter and a signal provided from the equalizing circuit for supplying the audio output signal to the speaker.

[0020] In an exemplary implementation, the audio processing device may be installed in an interior vehicle environment, such as inside a car cabin. In such a case, the signal analyzing circuit stores a table containing multiple frequencies of room modes determined by measurement in the vehicle environment. In this way, an audio system may be adapted to the particular circumstances of a car environment.

[0021] For example, in an initial stage for implementing the audio processing device, at least one calibration signal may be transmitted into a room environment, such as the car cabin, the calibration signal containing a predetermined frequency spectrum. A signal response, particularly a frequency response of the acoustic transfer function between speaker and listening position, is measured for determining frequencies of room modes. With this information, a table is stored containing the determined frequencies of room modes which are used, in an operating stage, for the activating and deactivating of the frequency characteristic correction processing.

Brief Description of the Drawings

[0022] The invention will now be described with reference to a detailed description of embodiments of the invention and accompanying drawings, in which

Fig. 1 shows schematically a room view of an exemplary room mode for a cubic room, as already discussed;

Fig. 2 shows an exemplary room mode measurement depicting different room modes and their decay over time, as discussed earlier;

Fig. 3 shows a basic block diagram of an embodiment of an audio processing device according to the invention;

Fig. 4 shows two tables which may be stored in an audio processing device according to Fig. 3 for adapting the frequency characteristic correction processing for a particular room mode frequency over time;

Fig. 5 shows an exemplary time diagram for two particular audio input signal frequencies and the associated activating and deactivating of the frequency characteristic correction processing over time for these audio input signals;

Fig. 6 shows a signal diagram of an exemplary sound signal of a drum depicting typical short "sound elements" of a drum sound;

Fig. 7 shows a signal diagram of an exemplary sound signal of a drum depicting sound and attack, duration and decay times of a drum pulse.

Detailed Description of Embodiments

[0023] Fig. 3 shows a basic block diagram of an embodiment of an audio processing device according to the invention. The audio processing device 1 comprises an input terminal 2 configured to receive an audio input signal 20 from an audio source which is not shown in Fig.

1. A buffer circuit 4 is coupled to the input terminal 2 for successively storing a predetermined number of samples of the audio input signal 20. For example, the buffer circuit 4 has a length which is capable to store samples of the audio input signal 20 during a period of time which is in the order of a room mode onset or decay time. A low pass filter 5 is coupled between the buffer circuit 4 and an equalizing circuit 7, wherein the output of the low pass filter 5 is providing an output signal to the equalizing circuit 7 containing only lower frequencies of the audio input signal, for example below a cut frequency Fc of 200 Hz. A high pass filter 6 is coupled in parallel to the series connection of the low pass filter 5 and the equalizing circuit 7.

[0024] The audio processing device further comprises an output circuit coupled to the output of the equalizing circuit 7 which is configured to supply an audio output signal 30 to a speaker (not shown) for audio output from the speaker. Particularly, the output circuit comprises a summation circuit 8 for adding a signal provided from the high pass filter 6 and a signal provided from the equalizing circuit 7 for supplying the audio output signal 30 to the speaker. A static equalizing circuit 9 may be connected downstream to the summation circuit 8 for static EQ compensation of the audio signal. The static equalizing circuit 9 does not compensate the audio signal at the frequencies of room modes, which EQ compensation is handled by the equalizing circuit 7.

[0025] The input of the equalizing circuit 7 is coupled to the input terminal 2 via the buffer circuit 4 and the low pass filter 5. The equalizing circuit 7 is configured to execute, for specific frequencies of room modes, frequency characteristic correction processing (referred to in this example as EQ compensation) for a signal received at the input. A signal analyzing circuit, which includes a dynamic analysis portion 11 and a control portion 12, is coupled to the input terminal 2 through the buffer circuit 4 and the low pass filter 5 to receive the audio input signal 20 via the buffer circuit 4 and the low pass filter 5. The signal analyzing circuit 11, 12 is configured to analyze a frequency characteristic, at least at signal components of lower frequencies, of the audio input signal 20 for detecting the presence of a signal power of the audio input signal 20 at or near one or more frequencies of a room mode. In this context, detecting the presence of a signal power of the audio input signal at or near one or more frequencies of a room mode means detecting of a particular signal power within a narrow frequency band around one or more room mode frequencies, such as F1, F2, as explained below. The signal analyzing circuit, particularly its control portion 12 is coupled to a control terminal of the equalizing circuit 7 for dynamically adapting the EQ compensation for specific frequencies of room modes over time, as described in more detail in the following.

[0026] In an initial stage for implementing the system and method according to the invention, a calibration signal may be transmitted into the particular room environment, such as the car cabin. The calibration signal contains a predetermined frequency spectrum and a signal response or frequency response of the acoustic transfer function between speaker and listening position is measured for determining frequencies of room modes. In other words, the car cabin (or any other room) will be measured and explored to find the frequencies of the main room modes present at the relevant listening positions in the car. This kind of information will be used as a priori knowledge for the functioning of the EQ compensation according to the principles of the invention.

[0027] A table of frequencies and associated decay times for each room mode will be determined. This is done by a separate measurement as outlined above. Similar measurements (so-called "Waterfall" measurements as discussed above and shown with reference to Fig. 2) should be done for determining the respective onset time of the room modes. Again, a table with frequencies and associated onset times will be determined. An example of such a table is shown in Fig. 4A depicting a table 21 in which the frequencies F1 to F3 and the corresponding onset times of exemplary principal room modes are stored. Another example of a table is shown in Fig. 4B depicting a table 22 in which the respective decay times of the exemplary principal room modes at the frequencies F1 to F3 are stored. Both tables 21 and 22, which may be stored in the dynamic analysis portion 11 or in a separate memory, are shown as separate tables, but may also be integrated as a common table. Depending on the particular room, the onset and decay time can vary for each room mode frequency. Decay time and onset time can also be the same, i.e. the room "behaves" in a symmetric way. For keeping generality two tables may be used, one for decay time and one for onset time.

[0028] In the following, the function of the audio processing device as shown in Fig. 3 will be explained with particular reference to Fig. 5, showing an exemplary time diagram for two particular audio input signal frequencies and the associated activating and deactivating of the frequency characteristic correction processing over time for these audio input signals.

[0029] The buffer circuit 4, functioning as input buffer, implements a signal analysis buffer of a certain length. The length of this buffer should be in the order of the decay or onset time of the room mode having the lowest frequency. The low pass filter 5 is designed as quadrature mirror filter (QMF) with a high pass filter (HPF). The cut frequency may be at 200 Hz, as room modes as a result of reflection at room walls or the like are typically present at lower frequencies. The high pass filter 6 is designed as QMF with a low pass filter (LPF). The cut frequency may also be at 200 Hz. The summation circuit 8 adds the signal coming from the output of the HPF 6 and the signal coming from the output of the equalizing circuit 7. The static equalizing circuit 9 ("Static EQ without Mode Compensation") implements the car audio equalization derived from static measurements (tuning) without considering room mode compensation. In the tuning process every frequency where a room mode is present is analyzed. Any negative cut necessary to achieve a desired target curve will be shifted to the equalizing circuit 7.

[0030] The equalizing circuit 7 ("EQ Cuts per Frequency Room Mode") comprises a series of cut filters at the room modes present in the car cabin (F1, F2, F3, etc., see Fig. 4) implemented as second order section IIR filters. The EQ control portion 12 of the signal analyzing circuit controls the equalizing circuit 7. The controlling will be activated by the output of the dynamic analysis portion 11 of the signal analyzing circuit. If the dynamic analysis portion is requesting action, the equalizing circuit 7 will filter the low frequency signal from the LPF 5 applying frequency cuts, i.e. compensating room modes. In other words, peaks of the signal components having a specific frequency of or near a room mode are cut in order to equalize, in connection with the static equalizing circuit 9, the frequency characteristic of the audio signal to a substantially flat curve over the frequency band (flat signal response or frequency response of the acoustic transfer function between speaker and listening position). If action is not required the equalizing circuit 7 will be gradually switched into a bypass filter (room modes are not compensated). The dynamic analysis portion 11 controls the time for activation and deactivation of the equalizing circuit 7. It performs frequency analysis over time to determine activation and deactivation of the equalizing circuit 7 via the EQ control portion 12.

[0031] The analysis of the frequency characteristic of the audio input signal 20 is functioning as follows:

[0032] The internal buffer circuit 4 holds a certain amount of samples enough to perform frequency analysis of the input signal 20. This buffer 4 will be actualized with an amount of samples incoming to the audio system (block based system) in a circular way. The outputting of the signal will be delayed in the buffer 4 in order to control short duration sound in the low frequency region of the spectrum. The content of the buffer 4 will be frequency transformed using FFT (Fast Fourier Transformation). The window size will be selected to have a reasonable trade off between frequency and time resolution.

[0033] Two kinds of detection are performed: time based and frequency based. The initial state of the equalizing circuit 7 will be an "all pass" on all frequencies (0 db for all frequencies). This filter will contain gain variable cutting filters (notch profile) at each room mode frequency, e.g. as listed in previously mentioned tables 21 and 22 of Fig. 4. Further, power or energy band detection will be performed by means of a narrow band filter at the frequencies where room modes appear (around F1, F2, F3, etc.). This means that a gain threshold will be compared to the current spectral energy or power on that band. Particularly, signal power or energy is estimated from several blocks in time from buffer 4 (previously FFT transformed) by means of averaging within a narrow frequency band around F1, F2. If the energy or power within the band is greater than the threshold the detection is positive (a room mode frequency was detected in the audio signal). The energy band detection is repeated periodically (in small blocks of samples) wherein the duration of the detection of the presence of the signal power to be determined is the time in samples or milliseconds that the input signal provides a positive power detection for or near a room mode frequency.

[0034] A different timer (e.g. counter) will be assigned for each individual room mode frequency, for example a timer1 for F1, timer2 for F2, timer3 for F3 and so on. Each timer will be started at a respective point of time (e.g. Ton (F1) for compensating a signal at frequency F1, Ton (F2) for compensating a signal at frequency F2) depending on the positive detection of its corresponding room mode frequency. The respective timer will be deactivated at a respective point of time (e.g. Toff (F1) for deactivating compensating at frequency F1, Toff (F2) for deactivating compensating at frequency F2) depending on the non-detection (low energy or not present) of its corresponding room mode frequency.

[0035] Each timer is activated after energy band detection is positive, i.e. the energy or power of the audio input signal for or at the respective room mode frequency (i.e., within the narrow frequency band around the respective room mode frequency) is greater than the threshold (i.e., a corresponding room mode frequency was detected in the audio signal). The respective timer continues counting only if the energy is over the gain threshold over time. If the positive detection lasts longer than the time threshold for F1, F2, etc., the EQ control portion 12 will be activated. At this time the corresponding timer will be reset to zero. This indicates that currently the signal will excite a room mode, therefore steady state cuts or EQ compensation for F1, F2 (designated as "Cut EQ" in Fig. 5) should be applied.

[0036] The EQ control portion may implement a "fade in" and "fade out" for the activation and deactivation of the cut filters of the equalizing circuit 7 correspondingly. Time and profile for this fade in and fade out are parameters of the EQ control portion 12. Fading will be implemented for changing the gain cut of the notch filters of the equalizing circuit 7. This gain control of the equalizing circuit's 7 parametric EQs will be performed in order to ensure a soft transition between the states "cutting EQ mode" and "bypass setting" during transition from deactivation to activation and vice versa.

[0037] If the signal detected lasted less than the time needed for the formation of a room mode, no frequency cut will be activated. This means the EQ control portion 12 will keep the equalizing circuit 7 to be a bypass filter. An example is a room mode onset time of 300 ms and the signal lasts 40 ms at F1 room mode (e.g. F1 = 40Hz). Depending on the parameters of the EQ control portion 12 it could be that during a fade in (activation) or fade out (deactivation) phase an opposite transition happens. For example, time for activation was achieved, fade in is in process and a transition to deactivation is detected. These special cases are shown in Fig. 5.

[0038] In Figures 6 and 7, examples of audio signals are shown which represent a drum music signal. A drum sound as shown is characteristical for having short lasting low frequency signals having relative high energy which may excite room modes if they last longer than the corresponding onset time. According to the invention, however, such drum audio signals are not subjected to EQ compensation, since the typical drum "sound elements" as shown in Fig. 6 are shorter than typical onset and decay times of the corresponding room modes. For example, in Fig. 7, the duration time of the drum "sound element" (short hit on the drum) is in the order of 0.04 ms. Since room modes will not be developed as a matter of the short duration of the corresponding signal power at that frequency, frequency characteristic correction processing (EQ compensation) is not necessary for these signal components and, thus, the equalizing circuit 7 is not activated for these signal components. Therefore, the typical short drum "sound elements" (also when mixed with other sound elements of other music instruments) pass through the equalizing circuits 7 and 9 without overcompensation as they are not subjected to the corresponding frequency cut filters, so that short high energy and low frequency sound is allowed to be reproduced without room mode compensation.

[0039] Although the invention has been described with particular reference to implementing a car audio system, this kind of technology is not restricted to the particular embodiments explained therein and may be used in various application fields of audio systems, such as home or theater entertainment systems, where a standard steady state equalization is used, supplemented with a dynamic compensation technology according to the invention especially for low frequency signals.

Claims

1. An audio processing device, comprising:

- an input terminal (2) configured to receive an audio input signal from an audio source,
- an equalizing circuit (7) having an input, an output and a control terminal, wherein the input is coupled to the input terminal (2) and the equalizing circuit (7) is configured to execute frequency characteristic correction processing for a signal received at the input,
- an output circuit (8, 9, 3) coupled to the output of the equalizing circuit (7) and configured to supply an audio output signal to a speaker for audio output from the speaker,
- a signal analyzing circuit (11, 12) coupled to receive the audio input signal and configured to analyze a frequency characteristic of the audio input signal for detecting the presence of a signal power of the audio input signal at at least one frequency (F1, F2) of a room mode,
- wherein the signal analyzing circuit (11, 12) is coupled to the control terminal of the equalizing circuit (7) for dynamically adapting the frequency characteristic correction processing of the equalizing circuit (7) at the at least one frequency of a room mode over time when the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode is detected.

2. The audio processing device of claim 1, wherein

- the signal analyzing circuit (11, 12) stores a predetermined onset time for the at least one frequency (F1, F2) of a room mode,
- if the signal analyzing circuit (11, 12) detects the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode, the signal analyzing circuit determines a duration of the detection of the presence of the signal power and compares the duration with the stored onset time,
- wherein if the duration is equal to or greater than a predetermined time threshold, the signal analyzing circuit (11, 12) controls the equalizing circuit (7) for activating the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode.

3. The audio processing device of claim 1 or 2, wherein

- the signal analyzing circuit (11, 12) further determines a signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode and compares the determined signal power with a signal power threshold value,
- wherein if the determined signal power is equal to or greater than the signal power threshold value, the signal analyzing circuit (11, 12) controls the equalizing circuit (7) for activating the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode.

4. The audio processing device of claim 2 or 3, wherein

- the signal analyzing circuit (11, 12) stores a predetermined decay time for at least one frequency (F1, F2) of a room mode,
- the signal analyzing circuit (11, 12) further determines a signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode and compares the determined signal power with a signal power threshold value,
- if the signal analyzing circuit (11, 12) has previously controlled the equalizing circuit (7) for activating the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode, the signal analyzing circuit

determines whether the determined signal power is lower than the signal power threshold value,
- and if the determined signal power is lower than the signal power threshold value, the signal analyzing circuit (11, 12) controls the equalizing circuit (7) for deactivating the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode after the decay time has elapsed counted from the point of time of when the signal analyzing circuit has determined that the determined signal power is lower than the signal power threshold value.

5. The audio processing device of one of claims 1 to 4, wherein
the signal analyzing circuit (11, 12) stores a table (21, 22) containing multiple frequencies (F1, F2, F3) of respective room modes, wherein the table associates a respective predetermined onset time and a respective predetermined decay time of the room modes with a corresponding one of the multiple frequencies.

6. The audio processing device of one of claims 1 to 5, further comprising
a buffer circuit (4) coupled between the input terminal (2) and the equalizing circuit (7) for successively storing a number of samples of the audio input signal, wherein the signal analyzing circuit (11, 12) is coupled to the buffer circuit (4) for analyzing a frequency characteristic of the audio input signal.

7. The audio processing device of claim 6, wherein
the buffer circuit (4) has a length which is capable to store samples of the audio input signal during a period of time which is in the order of a room mode decay or onset time.

8. The audio processing device of one of claims 1 to 7, further comprising
a low pass filter (5) coupled between the input terminal (2) and the equalizing circuit (7) and having an output providing an output signal to the equalizing circuit containing only lower frequencies of the audio input signal.

9. The audio processing device of claim 8, further comprising

- a high pass filter (6) coupled between the input terminal (2) and the output circuit (8, 9, 3) and in parallel to the series connection of the low pass filter (5) and the equalizing circuit (7),
- wherein the output circuit (8, 9, 3) comprises a summation circuit (8) for adding a signal provided from the high pass filter (6) and a signal provided from the equalizing circuit (7) for supplying the audio output signal to the speaker.

10. The audio processing device of one of claims 1 to 9, wherein
the audio processing device (1) is implemented in an interior vehicle environment and the signal analyzing circuit (11, 12) stores a table (21, 22) containing multiple frequencies (F1, F2, F3) of room modes determined by measurement in the vehicle environment.

11. A method of performing frequency characteristic correction processing for an audio input signal, comprising

- analyzing a frequency characteristic of an audio input signal (20) for detecting the presence of a signal power of the audio input signal at at least one frequency (F1, F2) of a room mode,
- performing, in an equalizing process, frequency characteristic correction processing for the audio input signal (20),
- dynamically activating and deactivating the frequency characteristic correction processing at the at least one frequency of a room mode over time if the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode has been detected,
- supplying from the frequency characteristic correction processing an audio output signal to a speaker for audio output from the speaker.

12. The method of claim 11, further including the steps of

- in an initial stage for implementing the method, transmitting at least one calibration signal into a room environment containing a predetermined frequency spectrum and measuring a signal response for determining frequencies (F1, F2) of room modes, and
- storing a table (21, 22) containing the determined frequencies (F1, F2) of room modes which are used, in an operating stage, for the activating and deactivating of the frequency characteristic correction processing.

13. The method of claim 11 or 12, further including the steps of,

- if the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode has been detected, determining a duration of the detection of the presence of the signal power and comparing the duration with an onset time of the corresponding room mode,
- if the duration is equal to or greater than a predetermined time threshold, activating the frequency characteristic correction processing at the at least one frequency (F1, F2) of the room

mode.

14. The method of claim 13, further including the steps of

- determining a signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode and comparing the determined signal power with a signal power threshold value,
- if the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode has been previously activated, determining whether the determined signal power is lower than the signal power threshold value,
- and if the determined signal power is lower than the signal power threshold value, deactivating the frequency characteristic correction processing at the at least one frequency (F1, F2) of a room mode after a decay time of the corresponding room mode counted from the point of time of when it has been determined that the signal power is lower than the signal power threshold value.

Amended claims in accordance with Rule 137(2) EPC.

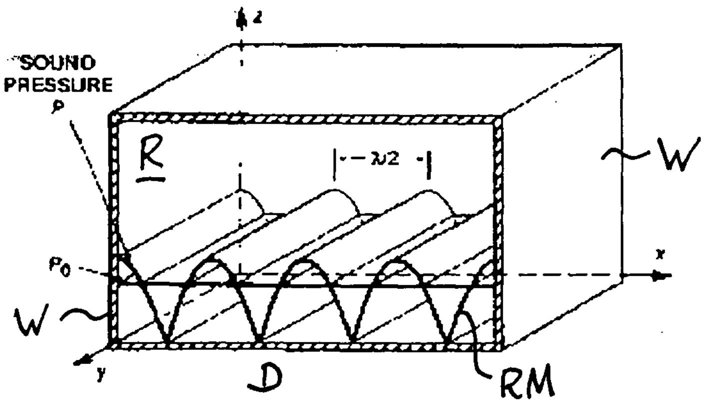
1. An audio processing device, comprising:

- an input terminal (2) configured to receive an audio input signal from an audio source,
- an equalizing circuit (7) having an input, an output and a control terminal, wherein the input is coupled to the input terminal (2) and the equalizing circuit (7) is configured to execute frequency characteristic correction processing for a signal received at the input,
- an output circuit (8, 9, 3) coupled to the output of the equalizing circuit (7) and configured to supply an audio output signal to a speaker for audio output from the speaker,
- a signal analyzing circuit (11, 12) coupled to receive the audio input signal and configured to analyze a frequency characteristic of the audio input signal for detecting the presence of a signal power of the audio input signal at at least one frequency (F1, F2) of a room mode,
- wherein the signal analyzing circuit (11, 12) is coupled to the control terminal of the equalizing circuit (7) for dynamically adapting the frequency characteristic correction processing of the equalizing circuit (7) at the at least one frequency of a room mode over time when the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode is detected, wherein the frequency characteristic correction processing is activated for the at least one frequency of a room mode if the detection of the presence of the signal power lasts longer than a time threshold for the at least one frequency of a room mode.

11. A method of performing frequency characteristic correction processing for an audio input signal, comprising

- analyzing a frequency characteristic of an audio input signal (20) for detecting the presence of a signal power of the audio input signal at at least one frequency (F1, F2) of a room mode,
- performing, in an equalizing process, frequency characteristic correction processing for the audio input signal (20),
- dynamically activating and deactivating the frequency characteristic correction processing at the at least one frequency of a room mode over time if the presence of the signal power of the audio input signal at the at least one frequency (F1, F2) of a room mode has been detected,
- wherein the frequency characteristic correction processing is activated for the at least one frequency of a room mode if the detection of the presence of the signal power lasts longer than a time threshold for the at least one frequency of a room mode,
- supplying from the frequency characteristic correction processing an audio output signal to a speaker for audio output from the speaker.

Fig. 1


SOUND
PRESSURE
p
R
λ/2
z
x
y
p_0
W
W
D
RM

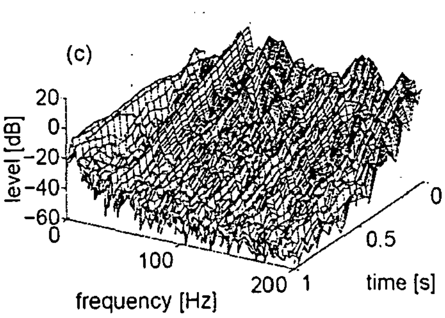

(c)
level [dB]
20
0
-20
-40
-60
0
100
200
frequency [Hz]
0
0.5
1
time [s]

Fig. 2

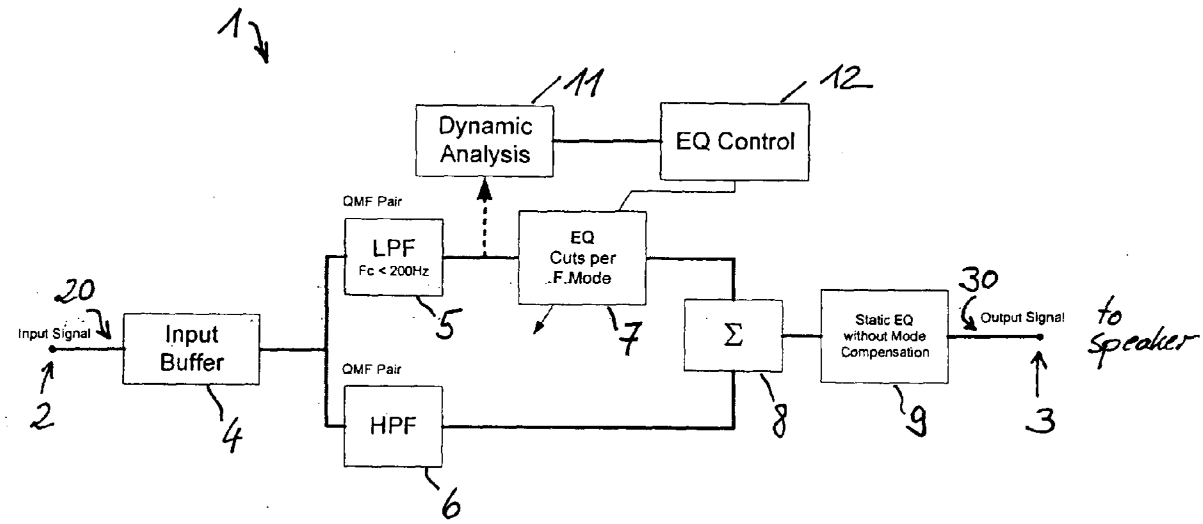


Fig. 3

A.

Frequency	On set time
F1 = 40 Hz	24 ms
F2 = 45 Hz	35 ms
F3 = 154 Hz	50 ms

21

B.

Frequency	Decay time
F1 = 40 Hz	22 ms
F2 = 45 Hz	36 ms
F3 = 154 Hz	45 ms

22

Fig. 4

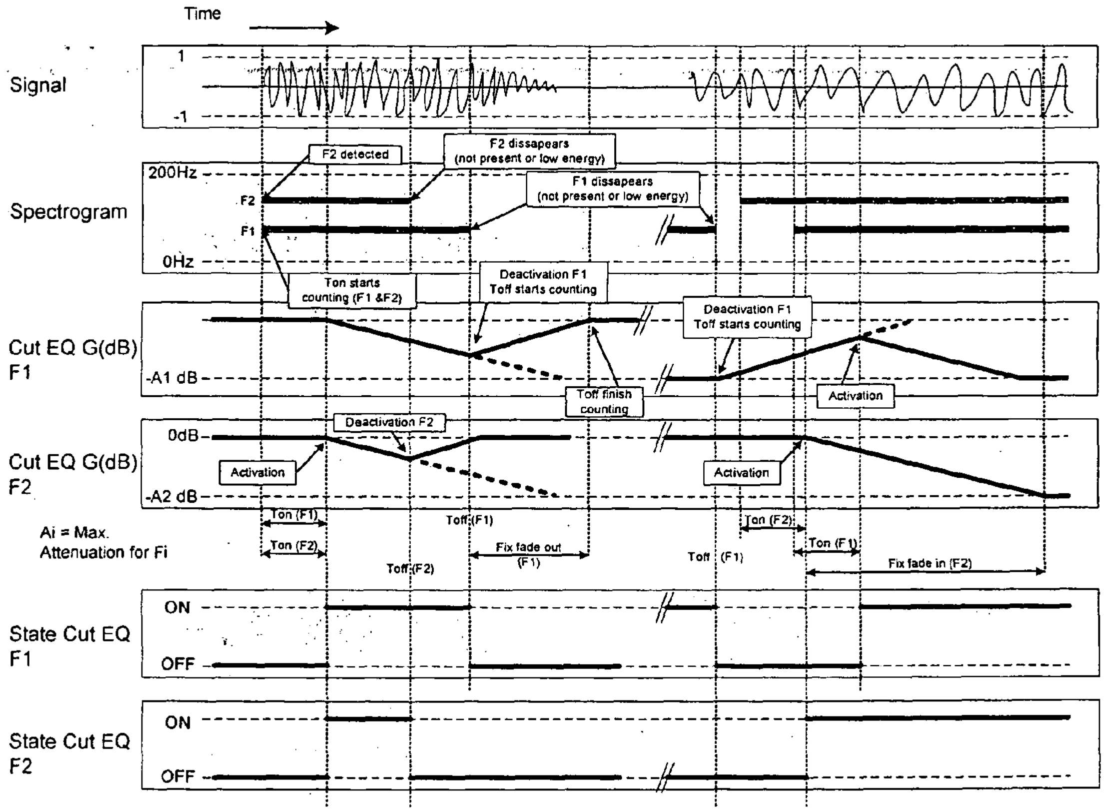

Time
Signal
1
-1
Spectrogram
200Hz
0Hz
F2
F1
F2 detected
F2 dissapears (not present or low energy)
F1 dissapears (not present or low energy)
Ton starts counting (F1 &F2)
Deactivation F1 Toff starts counting
Deactivation F1 Toff starts counting
Cut EQ G(dB) F1
-A1 dB
Toff finish counting
Activation
Deactivation F2
Cut EQ G(dB) F2
0dB
-A2 dB
Activation
Activation
Ai = Max. Attenuation for Fi
Ton (F1)
Ton (F2)
Toff (F1)
Toff (F2)
Fix fade out (F1)
Toff (F1)
Ton (F2)
Ton (F1)
Fix fade in (F2)
State Cut EQ F1
ON
OFF
State Cut EQ F2
ON
OFF

Fig. 5

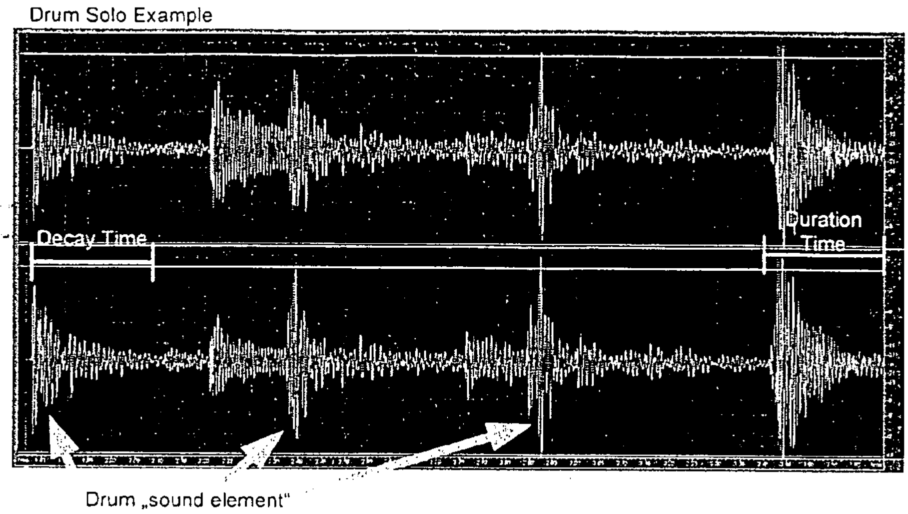

Drum Solo Example
Decay Time
Duration
Time
Drum „sound element"

Fig.6

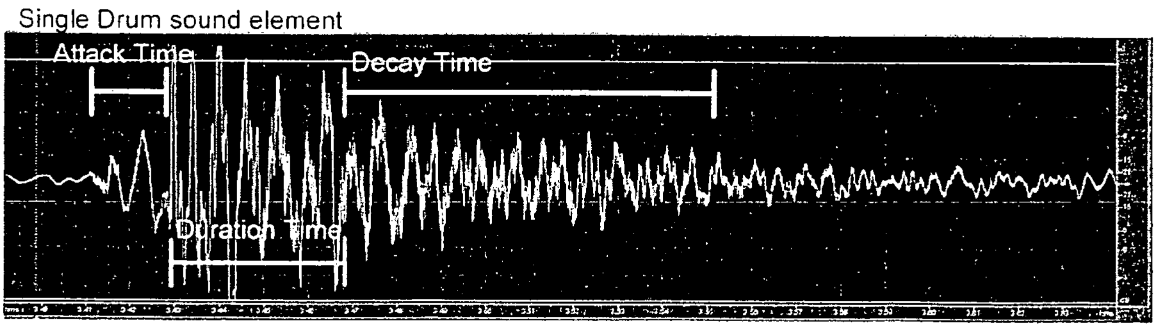

Single Drum sound element
Attack Time
Decay Time
Duration Time

Fig.7

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EUROPEAN SEARCH REPORT

Application Number
EP 08 01 0723

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
X	US 2004/002781 A1 (JOHNSON KEITH O [US]) 1 January 2004 (2004-01-01) * page 1, left-hand column, paragraph 4 * * page 2, left-hand column, paragraph 4 - page 6, left-hand column, paragraph 2 * * claims 1-13; figures 1-4 * -----	1,5-12	INV. H04R3/04 H04S7/00
A	WO 2007/068257 A (TC ELECTRONIC AS [DK]; CHRISTENSEN KNUD BANK [DK]) 21 June 2007 (2007-06-21) * page 56, line 6 - page 63, line 25 * * figures 4-6 * -----	1-14	
A	MAKIVIRTA A ET AL: "MODAL EQUALIZATION OF LOUDSPEAKER-ROOM RESPONSES AT LOW FREQUENCIES" JOURNAL OF THE AUDIO ENGINEERING SOCIETY, AUDIO ENGINEERING SOCIETY, NEW YORK, NY, US, vol. 51, no. 5, 1 May 2003 (2003-05-01), pages 324-343, XP001178321 ISSN: 1549-4950 * page 326, paragraph 1 - page 340, last paragraph * -----	1-14	TECHNICAL FIELDS SEARCHED (IPC) H04R H04S

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
Munich	28 November 2008	Meiser, Jürgen

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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EPO FORM 1503 03.82 (P04C01)

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The members are as contained in the European Patent Office EDP file on
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28-11-2008

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US 2004002781 A1	01-01-2004	NONE	
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Patent documents cited in the description

- JP 2000261900 A **[0008]**

Non-patent literature cited in the description

- **Poju Antsalo1.** Comparison of Modal Equalizer Design Methods. *AES Convention,* 22 March 2003, 5 **[0006]**