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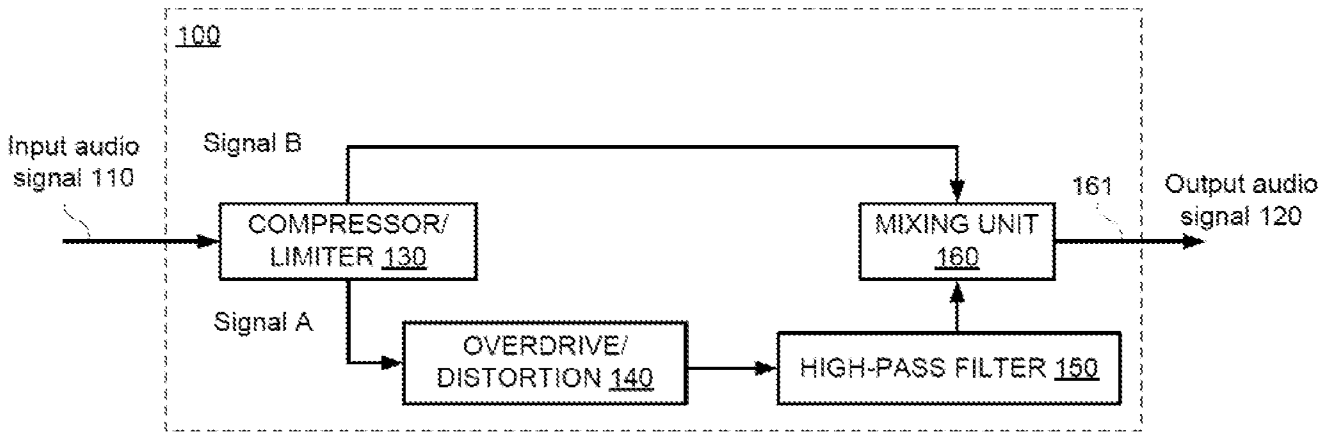
(54) Title **Method and device for processing an audio signal**

(56) References Cited: EP 3280155 A1, US 2017/0127731 A1, WO 2013/181299 A1

(57) Abstract

Method and device for processing an audio signal wherein the audio signal is compressed or limited before being split in first and second signals that may be separately processed to achieve desired effect(s) before the first and second signals are mixed with a desired or preferred balance.



Method and device for processing an audio signal

The present invention is related to a method for processing an audio signal, according to the preamble of claim 1.

The present invention is also related to an audio processing device, according to the preamble of claim 5.

Background

There exist many different prior art devices for modifying tonal qualities of electronic signals. In general, the prior art devices for processing musical instrument signals are very limited ability as regards providing the musician with a variety of desired tonal qualities in the resulting sound.

Presently, there exists no solutions that would allow a user to have full control and with the ability to affect one or more selections of frequencies in an audio signal to achieve desired effect(s) on the output signal.

The closest prior art for the present invention is known from US2017169808 A1, wherein is described an electronic signal processor for processing signals including a complex first filter, one or more gain stages and a second filter. The first filter is characterized by a frequency response curve that includes multiple corner frequencies, with some corner frequencies being user selectable. The first filter also has at least two user-preset gain levels, which may be alternately selected by a switch. Lower frequency signals are processed by the first filter with at least 12 db/octave slope, and preferably with 18 db/octave slope to minimize intermodulation distortion products by subsequent amplification in the gain stages. A second filter provides further filtering and amplitude control. The signal processor is particularly suited for processing audio frequency signals. Disadvantages with this solution is among others, that there is no compression or limiting of the signal. Further, there are no possibilities for splitting the signal and process one or both signals to achieve a desired effect of the output signal. There is further no possibility for overdrive or distortion, effect loop, low-pass/low-cut or high-pass/high-cut filtration, or octave effect on respective split signals.

In US5515446 A is disclosed an electronic audio processing system for insertion into the line prior to power amplification and a related method are disclosed which component and method render a frequency response in which undesirably strong frequency bands are flattened relative to the total frequency response by up to 10 dB, thereby providing enhanced dynamic audio reproduction. The audio signal is split into a plurality of frequency bands, each of which is actively, dynamically limited to a predetermined level related to the overall level of the unprocessed signal. The frequency bands are then recombined into a single processed signal which has an overall response, which is as much as 10 dB more linearity than the unprocessed signal. This is a multiband compressor/dynamic EQ with the purpose of leveling (flatten) the frequency response of a signal, which is the opposite goal of the present invention.

In EP 3280155 A1 is disclosed a signal processing device, signal processing method, and speaker device adapted for resolving volume shortage of middle and high band of speaker reproduction sound.

US 2017/0127731 A1 is related to selective audio signal enhancement, wherein an audio signal is enhanced by adding enhancement signals, such as harmonics and/or sub-harmonics of the signal's bass components. To avoid any unwanted enhancement signals during soft passages or speech, the audio signal is monitored for tonal signal components in a frequency range and adding only the enhancement signals if such tonal signals are detected.

From WO 2013/181299 A1 is known an effective and simple psychoacoustic bass generation system generating a harmonic signal having inter-modulation controllable to remain below a threshold level and includes a high-pass filter configured to pass harmonics which are reproducible with fidelity by the loudspeaker or other transducer and a loudness matching block configured to compensate the loudness of the desired harmonics to match the loudness of the original signal.

There is accordingly a need for a method and audio processing device providing compression or limiting of an audio signal that may be split into at least two signals that may be treated differently for achieving a desired effect on the output audio signal, something that will not be possible by using only one signal as is the case with the prior art solutions.

Object

The main object of the present invention is to provide a method for processing an audio signal and audio processing device partly or entirely solving the drawbacks of prior art solutions.

It is further an object of the present invention to provide a method for processing an audio signal and audio processing device enabling a user to achieve desired effect(s) on an audio signal.

An object of the present invention is to provide a method for processing an audio signal and audio processing device enabling a user to achieve desired effect(s) on selected frequencies of an audio signal.

It is an object of the present invention to provide a method for processing an audio signal and audio processing device capable of filtering out frequencies in a more efficient manner than prior art solutions are capable of.

Further objects will appear from the following description, claims and attached drawings.

The invention

A method for processing an audio signal according to the present invention is disclosed in claim 1. Preferable features of the method are disclosed in the dependent claims.

An audio processing device according to the present invention is disclosed in claim 5. Preferable features of the device are disclosed in the dependent claims.

The present invention is related to a novel and inventive way of processing audio signals. By the present invention is provided a method for processing audio signals and audio processing devices enabling a user to achieve desired effect(s) on an audio signal.

The method and audio processing device are based on performing a compression or limiting of a full input audio signal and splitting the compressed or limited signal into at least first and second signals that may be separately processed to achieve desired effect(s) on the audio signal, before the first and second signals are mixed according to a desired balance to provide a processed output signal.

According to the present invention there is performed an overdrive or distortion of the first signal before the first and second signals are mixed according to a desired balance.

In accordance with a further embodiment of the present invention, in addition to overdrive or distortion of the first signal, there is performed a low-pass filtering of the second signal before the first and second signals are mixed according to a desired balance.

According to a further embodiment of the present invention, the first signal, in addition to overdrive or distortion, is further high-pass filtered in front, after or both of the overdrive or distortion processing.

In accordance with a further embodiment of the present invention the first signal, in addition to overdrive or distortion, is applied an octave effect, in bypass over the overdrive or distortion processing or in series with the overdrive or distortion processing.

According to a further embodiment of the present invention the first signal, in addition to overdrive or distortion, is applied an effect loop, in bypass over the overdrive or distortion processing or in series with the overdrive or distortion processing.

In accordance with a further embodiment of the present invention, the first signal, in addition to overdrive or distortion, is high-cut filtered, after the overdrive or distortion processing, or that the mixed signal is high-cut filtered.

According to a further embodiment of the present invention the second signal, in addition to low-pass filtering, is low-cut filtered or that the mixed signal is low-cut filtered.

In accordance with a further embodiment of the present invention, there is performed equalization of the second signal or mixed signal.

According to a further embodiment of the present invention, the mixed signal is impulse response filtered.

A method for processing an audio signal accordingly comprises compressing or limiting an input audio signal, as well as splitting the compressed or limited audio signal into first and second signals. The method further comprises performing overdrive or distortion of the first signal and high-pass filtering the first signal before the first and second signals are mixed with a desired balance.

The method according to a further embodiment of the present invention further comprises performing or applying one or more of the following actions for the first signal before the first and second signals are mixed:

- high-pass filtering in front of the overdrive or distortion processing,

- adding an octave effect, in bypass over the overdrive or distortion processing or in series with the overdrive or distortion processing,
- adding an effect loop, in bypass over the overdrive or distortion processing, in series with the overdrive or distortion processing or in front of the overdrive or distortion processing, and/or
- high-cut filtering, after the overdrive or distortion processing.

The method according to a further embodiment of the present invention further comprising performing one or more of the following actions to the second signal before the first and second signals are mixed:

- low-pass filtering,
- low-cut filtering, and/or
- equalization.

The method according to a further embodiment of the present invention further comprising performing one or more of the following actions to the mixed first and second signal:

- high-cut filtering,
- low-cut filtering,
- equalization, and/or
- impulse response filtering.

Accordingly, in the present invention the input audio signal is compressed or limited and next split into first and second signals, which then may be separately processed to achieve desired effect(s) and wherein the separately processed signals are mixed with a desired balance to provide a processed output audio signal.

The present invention thus provides a method and device for processing an audio signal where the user may achieve desired effect(s) on selected frequencies of the audio signal and which next are balanced according to the desires of the user.

By the present invention is thus provided an analogue or digital multiband signal processing chain that provides opportunities for controlled overdrive or distortion and effects (pitch, dynamic, filter, modulation, spatial and time based), that may be separately controlled according to the desires of the user.

Further preferable features and advantageous details of the present invention will appear from the following example description, claims and attached drawings.

Example

The present invention will below be described in further detail with references to the attached drawings, where:

Fig. 1 is a principle drawing of a first embodiment of an audio processing device according to the present invention,

Fig. 2 is a principle drawing of a second embodiment of an audio processing device according to the present invention,

Fig. 3 is a principle drawing of a third embodiment of an audio processing device according to the present invention,

Fig. 4a-b are principle drawings of a fourth embodiment of an audio processing device according to the present invention,

Fig. 5a-c are principle drawings of a fifth embodiment of an audio processing device according to the present invention,

Fig. 6 is a principle drawing of a sixth embodiment of an audio processing device according to the present invention,

Fig. 7 is a principle drawing of a seventh embodiment of an audio processing device according to the present invention,

Fig. 8 is a principle drawing of an eight embodiment of an audio processing device according to the present invention,

Fig. 9 is a principle drawing of a ninth embodiment of an audio processing device according to the present invention, and

Fig. 10 is a principle drawing of a tenth embodiment of an audio processing device according to the present invention.

The present invention will below be described by reference to the Figures showing embodiments of the present invention by means of block diagrams. The block diagrams of the different embodiments make use of common blocks for the same technical features. An introduction of the blocks, as well as a definition thereof is given below.

Limiter: A limiter is per definition an audio limiter. A limiter is a type of compressor designed for a specific purpose, and in the present invention used to limit the level of a signal to a certain threshold. Whereas a compressor will begin smoothly reducing the gain above the threshold, a limiter will almost completely prevent any additional gain above the threshold. Controllable parameters of the limiter will be threshold, release and gain.

Compressor: A compressor performs dynamic range compression (DRC) or simply compression. Compression is an audio signal processing operation that reduces the volume of loud sounds or amplifies quiet sounds thus reducing or compressing an audio signal's dynamic range. Controllable parameters of the compressor will be threshold, ratio, attack, release, gain, as wells a hard and soft knee selector.

Low-pass filter: A low-pass filter (LPF) is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The amount of attenuation for each frequency depends on the filter design. The filter is sometimes called a high-cut filter, or treble-cut filter in audio applications. A low-pass filter is the complement of a high-pass filter. Controllable parameters of the low-pass filter will be cut-off frequency and slope.

High-pass filter: A high-pass filter (HPF) is an electronic filter that passes signals with a frequency higher than a certain cutoff frequency and attenuates signals with frequencies lower than the cutoff frequency. The filter is sometimes called a low-cut filter in audio applications. The amount of attenuation for each frequency depends on the filter design. Controllable parameters of the High-pass filter will be cut-off frequency and slope.

Equalizer: Equalization, or EQ for short, means boosting or reducing (attenuating) the levels of different frequencies in a signal. The most basic type of equalization familiar to most people is the treble/bass control on home audio equipment. The treble control adjusts high frequencies, the bass control adjusts low frequencies. The EQ can have several bands and several shapes (High-pass, Low-pass, High shelf, Low shelf, Bell shape). The controllable parameters of the equalizer will be, among others, Parametric EQ, Gain (Per band), Center frequency (Per band), Band with (Q) (per band), Filter shape selector (per band), Graphic EQ, Gain (per band).

Impulse response filter: Impulse response (commonly known as IR) is a frequency response wav file, often captured from a setup with a microphone capturing the frequency response of a speaker (and its environment). The impulse response filter can then be used to pass audio through, filtering the signal to have the same frequency response as the mic and speaker used to capture the IR had. The controllable parameters of the impulse response filter will be Blend/mix between two or more IR´s, volume for each IR.

Distortion/overdrive: Distortion and overdrive are forms of audio signal processing used to alter the sound of amplified electric musical instruments, usually by increasing their gain, producing a "fuzzy", "growling", or "gritty" tone. The controllable parameters of the distortion/overdrive will be: Drive amount (Gain), type of distortion selector, bypass.

Octave effect/Octaver: An octave effect takes the note (signal) that it detects, and shifts it by exactly 12 notes either up or down. This means that if you play a C note through an octave effect, you will still hear a C, but it will be either an octave higher or lower (or both) than you played it. The controllable parameters of the octave effect/octaver will be Octave up/down selector (Toggle switch), on/off selector for octave up and octave down, octave volume for the selected octave or separate volume for each of the octave.

Effects loop: An effects loop is one or more audio effects units, connected between two points of a signal path (the route that a signal would travel from the input to the output); usually between the pre-amp and power amp stages of an amplifier circuit. The controllable parameters of the effect loop will be On/Off switch.

Reference is now made to Fig. 1, which is a principle drawing of a first embodiment of an audio processing device 100 for processing an audio signal according to the present invention. The audio processing device 100 according to the present invention is provided with an input for an audio signal 110 to be processed by the audio processing device 100 and an output 120 for output of the processed audio signal from the audio processing device 100.

The audio processing device 100 according to the present invention is further provided with a controllable compressor or limiter 130 connected to the input receiving the input audio signal 110. The controllable compressor or limiter 130 compresses or limits the full audio signal 110 and is further arranged to split the processed signal into first A and second B signal paths of the same signal. Alternatively, the audio processing device 100 comprises a separate splitter (not shown) arranged to the output of the compressor or limiter 130 for splitting the signal after the signal has been processed by the compressor or limiter 130. According to an alternative embodiment of the present invention the controllable compressor or limiter 130 is arranged external of the audio processing device 100, i.e. in front of the input of the audio processing device 100, and wherein the input signal is arranged to a separate splitter (not shown) in the audio processing device 100.

The audio processing device 100 is further comprising means for processing the first signal A in the form of a controllable overdrive or distortion unit 140 arranged in the first signal A path, in the shown embodiment receiving the first signal A from the compressor or limiter 130, and at least one controllable or fixed high-pass filter 150 arranged in the first signal A path, after the controllable overdrive or distortion unit 140, resulting in a processed first signal A at the output of the at least one controllable or fixed high-pass filter 150. The controllable or fixed high pass filter 150 will e.g. be an n^{th} order high-pass filter, where n will be between, but not limited to, 1-5, and e.g. have a frequency range between, but not limited to, 100-5000 Hz.

The audio processing device 100 is further comprising a mixer or mixing unit 160 that combines the first A and second B signal A to form the processed audio signal 120 as output of the audio processing device 100. The mixing unit 160 is e.g. controlled by a potentiometer or volume control for selecting the balance between the first A and second B signal.

By means of the compressor or limiter 130 the input audio signal 110 is compressed and split into the first A and second B one may process and affect differently. In this embodiment the second signal B is not further processed, while the first signal A is processed through the overdrive or distortion unit 140 where drive amount (gain) and type of distortion is added to the first signal A before the signal is further processed by the at least one fixed or controllable high-pass filter 150 to filter out frequencies higher than the set frequency range of the at least one high-pass filter 150. In this manner one can add the effect of drive amount and type of distortion to the first signal A, before the first A and second B signal are mixed by a preferred/desired balance by the mixing unit 160. I.e. the embodiment provides a solution wherein frequency signals in the mid-range of the input signal are affected. By preferred/desired balance is meant that the balance between the first signal A and the second signal B is given by e.g. X first signal A + Y second signal B = Z, wherein the X and Y are positive numbers e.g. between 0.1% and 99,9% and Z = 100 %, or X and Y and integer numbers between 1 and 99. Another scale that may be used for balancing between the first and second signal may be that X and Y are positive numbers between 0 and 0.99 and Z=1. Other scales for controlling the balance between the first A and second B signal will be obvious for a skilled person, based on these examples, such as logarithmic scales, S-curve-scale, or similar. Accordingly, if the balance between the first signal A and second signal B is 50%/0.5, the mixed signal 161 will be 50 %/0.5 of the first signal A and 50 %/0.5 of the second signal B. If e.g. the balance between the first signal A and second B signal at the ratio 70%/0.7 and 30 %/0.3, the mixed signal 161 will be 70%/0.7 of the first signal A and 30 %/0.3 of the second signal B. The number examples are just for illustration purposes and not to be considered as limiting for balance of the signals in the present invention, as other balances may also be used in the present invention.

In an alternative embodiment, not shown, there are arranged separate controls for controlling the amount of the respective first A and second B signals to the mixing unit 160. This will enable separately adjustable output volumes for the signal paths.

Reference is now made to Figure 2 which is a principle drawing of a second embodiment of the audio processing device 100 according to the present invention, which comprises the components of the first embodiment, but in addition comprises at least one fixed or controllable low-pass filter 170 arranged between the controllable compressor or limiter 130 and mixing unit 160 in the second signal B path. The at least one fixed or controllable low-pass filter 170 will e.g. be an n^{th} order low-pass filter, where n will be between, but not limited to, 1-5, and e.g. have a frequency range between, but not limited to, 60-1000 Hz.

According to the present invention, the low-pass filter(s) 170 and high-pass filter(s) 150 shall not overlap each other. If the low-pass filter 170 is set to filter out all frequencies up to a frequency f1, the high-pass filter(s) 150 will be set to filter frequencies from f2-f3, wherein f2>f1, and f3 is an upper limit. E.g. if the low-pass filter(s) 170 is set to filter out frequencies f1 lower than 200 Hz, the high-pass filter(s) 150 is set to filter out frequencies from f2=200 Hz to f3 of a desired value, such as e.g. 1 kHz.

In this embodiment, the first signal A is processed as described above, while the second signal B is processed through the at least one fixed or controllable low-pass filter 170 that filters out all frequencies in the second signal B below the set frequency of the low-pass filter 170 before the first A and second B signals are mixed in the mixing unit 160. This embodiment provides improved separation of the frequency bands comprising signal A and signal B. This reduces the additionality of the upper frequencies in the mix, and ensures better phase coherence.

Reference is now made to Figure 3 which is a principle drawing of a third embodiment of the audio processing device 100 according to the present invention, which comprises the components of the second embodiment, but in addition comprises at least one second fixed or controllable high-pass filter 151 arranged between the controllable compressor or limiter 130 and the controllable overdrive or distortion unit 140 in first signal A path. The second high-pass filter 151 could have the same settings as the high-pass filter 150 or a frequency range f4-f5, where the frequency f4=f2 and the frequency f5 is higher than the frequency f3. The controllable or fixed high-pass filter 151 will e.g. be an n^{th} order high-pass filter, where n will be between, but not limited to, 1-5, and e.g. have a frequency range between, but not limited to, 100-5000 Hz.

In this embodiment, the first signal A is processed by at least one fixed or controllable high-pass filter 151 filtering all frequencies above the mentioned low-pass filter 170, such that the first signal A to be processed is in a frequency range above the second signal B, which is processed by the overdrive or distortion unit 140, before the processed signal from the overdrive or distortion unit 140 is processed by at least one high-pass filter 150 with a desired frequency range. Accordingly, in this embodiment the overdrive or distortion unit 140 will only process frequencies higher than the frequencies of the processed second signal B.

This affects the tone of the distortion by filtering the frequencies to be processed by the overdrive or distortion unit 140.

Reference is now made to Fig. 4a-b showing a fourth embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second or third embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises at least one octave effect unit (octaver) 180 in the first signal A path. According to one embodiment of the present invention the octave effect unit 180 is arranged in a bypass of the overdrive or distortion unit 140, as shown in Figures 4a-b. Accordingly, the octave effect unit 180 receives the first signal A directly from the compressor or limiter 130 or the signal from the second high pass-filter 151, if present, and the signal from octave effect unit 180 is added to (summarized with) the signal of the overdrive or distortion unit 140 before the signal is supplied the high-pass filter 150, as shown in Figures 4a.

In an alternative embodiment, as shown in Fig. 4b, the octave effect unit 180 is arranged to the output of the overdrive or distortion unit 140, and the output of the octave effect unit 180 is added to (summarized with) the output signal of the overdrive or distortion unit 140 before the signal is supplied to the high-pass filter 150.

Accordingly, in this embodiment one in addition is processing the first signal A by an octave effect unit 180, either directly from the compressor or limiter 130 or via the high-pass filter 151, if present, bypassing the overdrive or distortion unit 140 or an octave effect unit 180 arranged to the output of the overdrive or distortion unit 140, wherein the output signal from the octave effect unit 180 is added to (summarized with) the output of the overdrive or distortion unit 140 before the signal is supplied to the high-pass filter 150. By this embodiment, the first signal A may be shifted an octave up or down to provide an effect on the processed output signal.

Accordingly, adding either a distorted octave effect (Fig. 4a), or a non-distorted octave effect (Fig. 4b) to the signal path A.

Reference is now made to Fig. 5a-c showing a fifth embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second, third or fourth embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises at least one effect loop 190 in the first signal A path. According to one embodiment of the present invention, the effect loop 190 is arranged in a bypass of the overdrive or distortion unit 140, as shown in Fig. 5a. Accordingly, the effect loop 190 receives the first signal A directly from the compressor or limiter 130 or the signal from the second high pass-filter 151, if present, and the signal from the effect loop 190 is added to (summarized with) the output of the overdrive or distortion unit 140.

In an alternative embodiment, as shown in Fig. 5b, the effect loop 190 is arranged in series between the overdrive or distortion unit 140 and the high-pass filter 150, accordingly processing the signal from the overdrive or distortion unit 140 before supplying it to the high-pass filter 150.

In an alternative embodiment, as shown in Fig. 5c, the effect loop 190 is arranged in series between the compressor or limiter 130 or the signal from the second high-pass filter 151, if present, and the overdrive or distortion unit 140.

Accordingly, in this embodiment the first signal A is in addition processed by an effect loop 190, either directly from the compressor or limiter 130 or via the high-pass filter 151, if present, and bypassing the overdrive or distortion unit 140 or arranged in series in front of or after the overdrive distortion unit 140.

This provides the possibility of adding third party effects to the signal chain of signal A.

Reference is now made to Fig. 6 showing a sixth embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second, third, fourth or fifth embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises a high-cut filter 200 in the first signal A path or in the mixed signal 161 from the mixing unit 160. According to the present invention, the high-cut filter 200 will be arranged in the first signal A path after the overdrive or distortion unit 140 or after the mixing unit 160, as shown in Fig. 6. The high-cut filter 200 is set to filter out frequencies higher than the high-pass filter(s) 150-151.

Accordingly, Figures 1, 3, 4a-b, 5a-c and 6 are disclosing how the first signal A may be processed. In the following there will be described how the second signal B may be processed according to the present invention.

Reference is now made to Figure 7 showing a seventh embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second, third, fourth, fifth or sixth embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises a low-cut filter 210 in the second signal B path or in the mixed signal 161 from the mixing unit 160. According to the present invention, the low-cut filter 210 may be arranged at any location in the second signal B path or after the mixing unit 160, as shown in Fig. 7. The low-cut filter 210 is set to filter out frequencies lower than the low-pass filter(s) 170.

Reference is now made to Figures 8 showing an eighth embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second, third, fourth, fifth, sixth or seventh embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises an equalizer 220 in the second signal B path or in the mixed signal 161 from the mixing unit 160. According to the present invention, the equalizer 220 may be arranged at any location after the low-pass filter 170 in the second signal B path or after the mixing unit 160, as shown in Fig. 7, alternatively without any low-pass filter 170 in the second signal B path.

Reference is now made to Figure 9 showing a ninth embodiment of the audio processing device 100 according to the present invention, which comprises at least the components of the first, second, third, fourth, fifth, sixth, seventh or eighth embodiment. In the shown embodiment, which is based on the embodiment of Fig. 3, the audio processing device 100 further comprises at least one impulse response filter 230 in the processed output audio signal 120. According to the present invention, the at least one impulse response filter 230 may be arranged at any location after the mixing unit 160 and applied to the mixed signal 161. The impulse response filter 230 is preferably arranged in front of the equalizer 220, if present.

The addition of an impulse response filter 230 gives the possibility to mimic speakers, cabinets or other acoustic elements.

Reference is now made to Figure 10 which is a principle drawing of a tenth embodiment where all the above described components have been included in the same embodiment of the audio processing device 100 according to the present invention by one of the above described embodiments. In addition, the audio processing device 100 comprises a gain reduction (GR) meter 300 that visually indicates the gain reduction applied by the compressor or limiter 130 for the user.

How the present invention works will now be described. By means of the compressor or limiter 130 the input audio signal 110 is compressed and split into first A and second B signals one may process differently. By means of the present invention, it is possible to process the first A and second B signals separately and combined to have effects on selected frequencies and filter out frequencies in a way that cannot be done on a single, full-range signal.

By the present invention it is possible to it is possible to selectively filter out undesired frequencies, such as e.g. the mid frequencies, in a way that is not possible with a bell shaped equalizer nor is it possible on a single signal, full-range signal.

The present invention provides opportunities for achieving desired effect(s) on a selected range of frequencies, and not on the full signal. E.g., one may achieve desired effect(s) on the high frequencies and not the low frequencies.

By the present invention, it is possible to have controlled overdrive or distortion on a selected frequency range of the audio signal. E.g., it will with the present invention be possible to have overdrive or distortion on the high frequency range of the audio signal and not on the low frequency range of the audio signal.

It is further possible to have controlled effects, such as pitch, dynamic, filter, modulation, spatial and time based, on the selected frequency range of the audio signal.

By the present invention, it is further possible to use an effect loop with external and/or integrated effects on a selected frequency range of the audio signal.

Further, as the mixing unit is controllable one will be able to control the balancing of the first and second signal as desired in relation to the application.

The present invention will especially be applicable for tonal audio signals, such as from music instruments, but may be used in any application where it is a need for processing the audio signal.

The above-described embodiments may be combined to form new embodiments within the scope of the claims.

Claims

1. A method for processing an audio signal, wherein the method comprises:

- compressing or limiting an input audio signal (110), as well as splitting the compressed or limited audio signal into a first (A) and second (B) signals,

- performing an overdrive or distortion of the first (A) signal,

- performing high-pass filtering after the overdrive or distortion processing, and

- mixing the first (A) and second (B) signals with a desired or preferred balance.

2. Method according to claim 1, wherein the method further comprising performing or applying one or more of the following actions to the first signal (A) before the first (A) and second (B) signals are mixed:

- high-pass filtering in front of the overdrive or distortion processing,
- adding an octave effect, in bypass over the overdrive or distortion processing or in series with the overdrive or distortion processing,
- adding an effect loop, in bypass over the overdrive or distortion processing, in series with the overdrive or distortion processing or in front of the overdrive or distortion processing, and/or
- high-cut filtering, after the overdrive or distortion processing.

3. Method according to claim 1 or 2, wherein the method further comprising performing one or more of the following actions to the second signal (B) before the first (A) and second (B) signals are mixed:

- low-pass filtering,
- low-cut filtering, and/or
- equalization.

4. Method according to any of the preceding claims, wherein the method further comprising performing one or more of the following actions to the mixed first (A) and second (B) signal:

- high-cut filtering,
- low-cut filtering,

- equalization, and/or
- impulse response filtering.

5. Audio processing device (100) comprising an input for an audio signal (110) and an output for a processed audio signal (120), wherein the audio processing device (100) comprising:

- an internal or external controllable compressor or limiter (130) for compressing or limiting an input audio signal, as well as arranged to split the compressed or limited audio signal into first (A) and second (B) signal paths, or split by a separate splitter arranged to the controllable compressor or limiter (130),

- a controllable overdrive or distortion unit (140) arranged in the first (A) signal path processing the first (A) signal,

- at least one controllable or fixed high-pass filter (150) in the first (A) signal path, after the controllable overdrive or distortion unit (140), and

- a mixing unit (160) mixing the first (A) and second (B) signal with a desired or preferred balance.

6. Audio processing device (100) according to claim 5, wherein the audio processing device (100) further comprises at least one second controllable or fixed high-pass filter (151) in the first (A) signal path, in front of the controllable overdrive or distortion unit (140).

7. Audio processing device (100) according to claim 5, wherein the audio processing device (100) further comprises at least one fixed or controllable low-pass filter (170) in the second (B) signal path.

8. Audio processing device (100) according to claim 5, wherein the audio processing device (100) further comprises at least one octave effect unit (180) in the first signal (A) path.

9. Audio processing device (100) according to claim 5, wherein the audio processing device (100) further comprises at least one effect loop (190) in the first signal (A) path.

10. Audio processing device (100) according to any preceding claim 5-9, wherein the audio processing device (100) further comprises one or more of:

- at least one high-cut filter (200) in the first signal (A) path arranged after the overdrive or distortion unit (140), or in mixed signal (161) from the mixing unit (160),

- at least one low-cut filter (210) in the second signal (B) path or in mixed signal (161) from the mixing unit (160), and/or

- at least one equalizer (220) in the second signal (B) path or in mixed signal (161) from the mixing unit (160).

Patentkrav

1. En fremgangsmåte for prosessering av et audiosignal, hvori fremgangsmåten omfatter:

- komprimering eller begrensning av et inngangsaudiosignal (110), samt splitting av det komprimerte eller begrensede audiosignalet opp i første (A) og andre (B) signaler,

- utføre en overstyring eller forvrenging av det første (A) signalet,

- utføre høypassfiltrering etter overstyrings- eller forvrengningsprosesseringen, og

- blande de første (A) og andre (B) signalene med en ønsket eller foretrukket balanse.

2. Fremgangsmåte i samsvar med krav 1, hvori fremgangsmåten videre omfatter å utføre eller anvende en eller flere av følgende aksjoner på det første signalet (A) før de første (A) og andre (B) signalene blandes:

- høypassfiltrering i front av overstyrings- eller forvrengingsprosesseringen,
- legge til en oktaveffekt, i bypass over overstyrings- eller forvrengingsprosesseringen eller i serie med overstyrings- eller forvrengingsprosesseringen,
- legge til en effektsløyfe, i bypass over overstyrings- eller forvrengingsprosesseringen, i serie med overstyrings- eller forvrengingsprosesseringen eller i front av overstyrings- eller forvrengingsprosesseringen, og/eller
- høykuttfiltrering, etter overstyrings- eller forvrengingsprosesseringen.

3. Fremgangsmåte i samsvar med krav 1 eller 2, hvori fremgangsmåten omfatter å utføre en eller flere av følgende aksjoner på det andre signalet (B) før de første (A) og andre (B) signalene blandes:

- lavpassfiltrering,
- lavkuttfiltrering, og/eller
- utjevning.

4. Fremgangsmåte i samsvar med ett av de foregående krav, hvori fremgangsmåten videre omfatter utføring av en eller flere av følgende aksjoner på det blandede første (A) og andre (B) signalet:

- høykuttfiltrering,
- lavkuttfiltrering,
- utjevning, og/eller

- impulsresponsfiltrering.

5. Audioprosesseringsanordning (100) omfattende en inngang for et audiosignal (110) og en utgang for et prosessert audiosignal (120), hvori audioprosesseringsanordningen (100) omfatter:

- en intern eller ekstern styrbar kompressor eller begrenser (130) for komprimering eller begrensning av et inngangsaudiosignal, samt innrettet til å splitte det komprimerte eller begrensede signalet opp i første (A) og andre (B) signalbaner, eller splitte med en separate splitter innrettet til den styrbare kompressoren eller begrenseren (130),

- en styrbar overstyrings- eller forvrengingsenhet (140) innrettet i den første (A) signalbanen for prosessering av det første (A) signalet,

- minst ett styrbart eller fast høypassfilter (150) i den første (A) signalbanen, etter den styrbare overstyrings- eller forvrengingsenheten (140), og

- en blandingsenhet (160) for blanding av det første (A) og andre (B) signalet med en ønsket eller foretrukket balanse.

6. Audioprosesseringsanordning (100) i samsvar med krav 5, hvori audioprosesseringsanordningen (100) omfatter minst ett andre styrbart eller fast høypassfilter (151) i den første (A) signalbanen, i front av den styrbare overstyrings- eller forvrengingsenheten (140).

7. Audioprosesseringsanordning (100) i samsvar med krav 5, hvori audioprosesseringsanordningen (100) videre omfatter minst ett fast eller styrbart lavpassfilter (170) i den andre (B) signalbanen.

8. Audioprosesseringsanordning (100) i samsvar med krav 5, hvori audioprosesseringsanordningen (100) omfatter minst en oktaveffektenhet (180) i den første (A) signalbanen.

9. Audioprosesseringsanordning (100) i samsvar med krav 5, hvori audioprosesseringsanordningen (100) omfatter minst en effektsløyfe (190) i den første (A) signalbanen.

10. Audioprosesseringsanordning (100) i samsvar med ett av kravene 5-9, hvori audioprosesseringsanordningen (100) omfatter en eller flere av:

- minst ett høykuttfilter (200) i den første (A) signalbanen innrettet etter overstyrings- eller forvrengingsenheten (140), eller i blandet signal (161) fra blandingsenheten (160),

- minst ett lavkuttfilter (210) i den andre (B) signalbanen eller i blandet signal (161) fra blandingsenheten (160), og eller

- minst en utjevningsenhet (220) i den andre (B) signalbanen eller blandet signal (161) fra blandingsenheten (160).

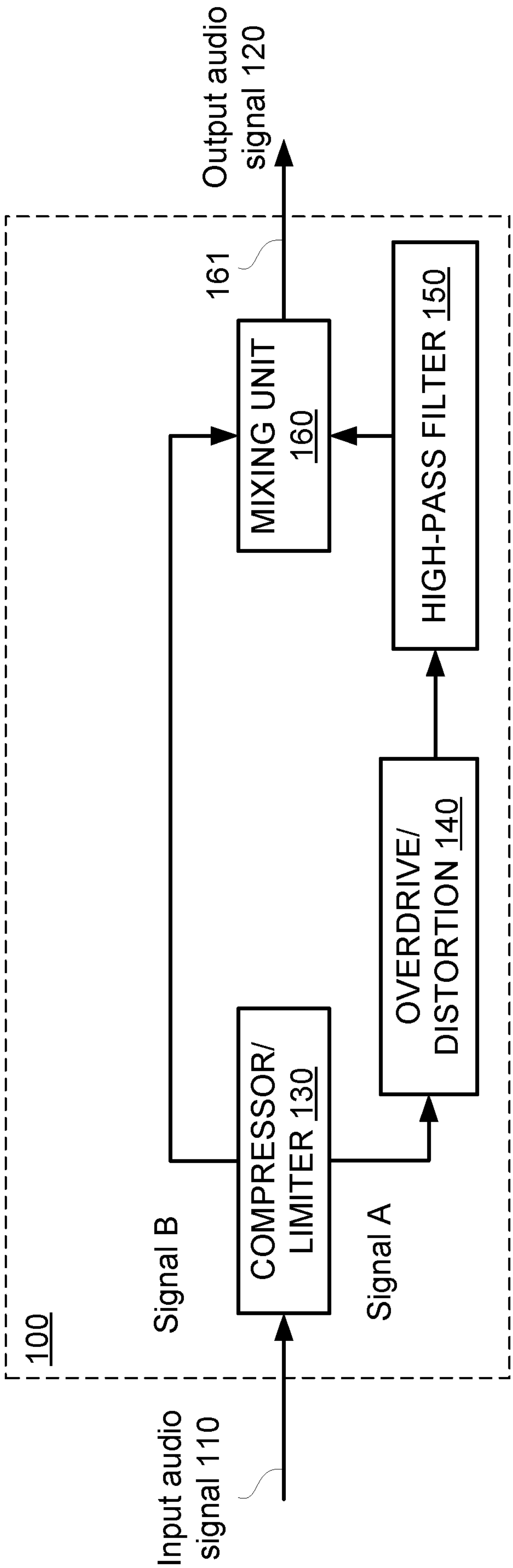

Output audio signal 120
161
MIXING UNIT 160
HIGH-PASS FILTER 150
OVERDRIVE/ DISTORTION 140
COMPRESSOR/ LIMITER 130
Signal B
Signal A
100
Input audio signal 110

Fig. 1.

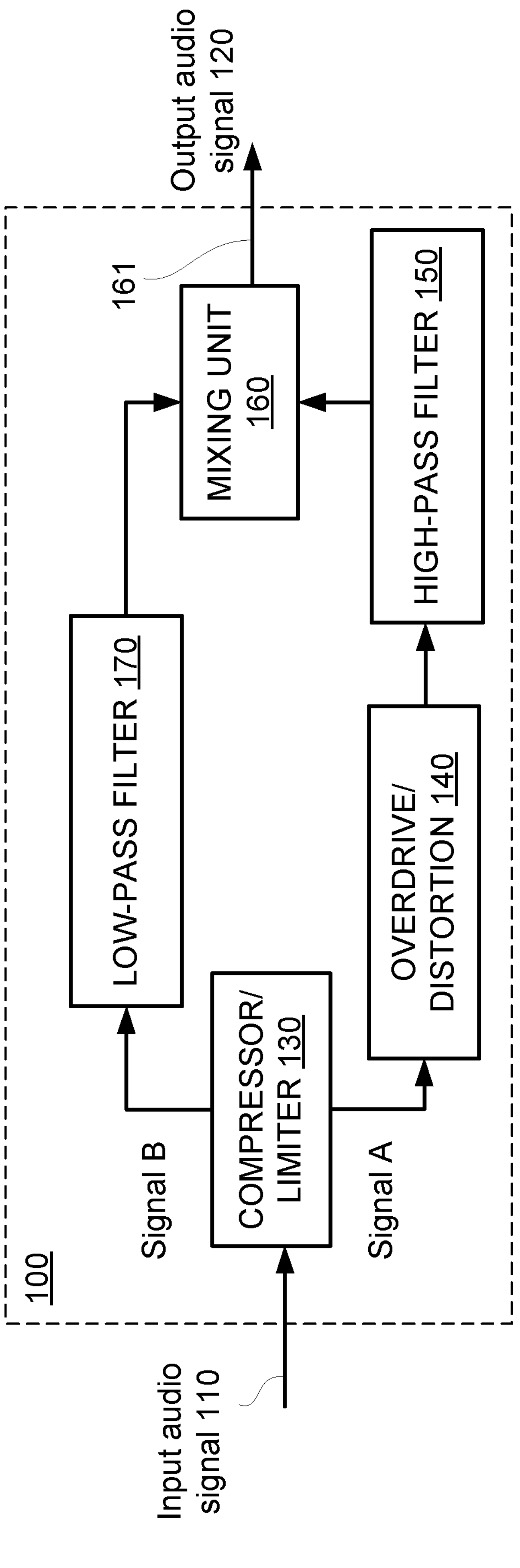

Output audio signal 120
161
MIXING UNIT 160
HIGH-PASS FILTER 150
LOW-PASS FILTER 170
OVERDRIVE/ DISTORTION 140
COMPRESSOR/ LIMITER 130
Signal B
Signal A
100
Input audio signal 110

Fig. 2.

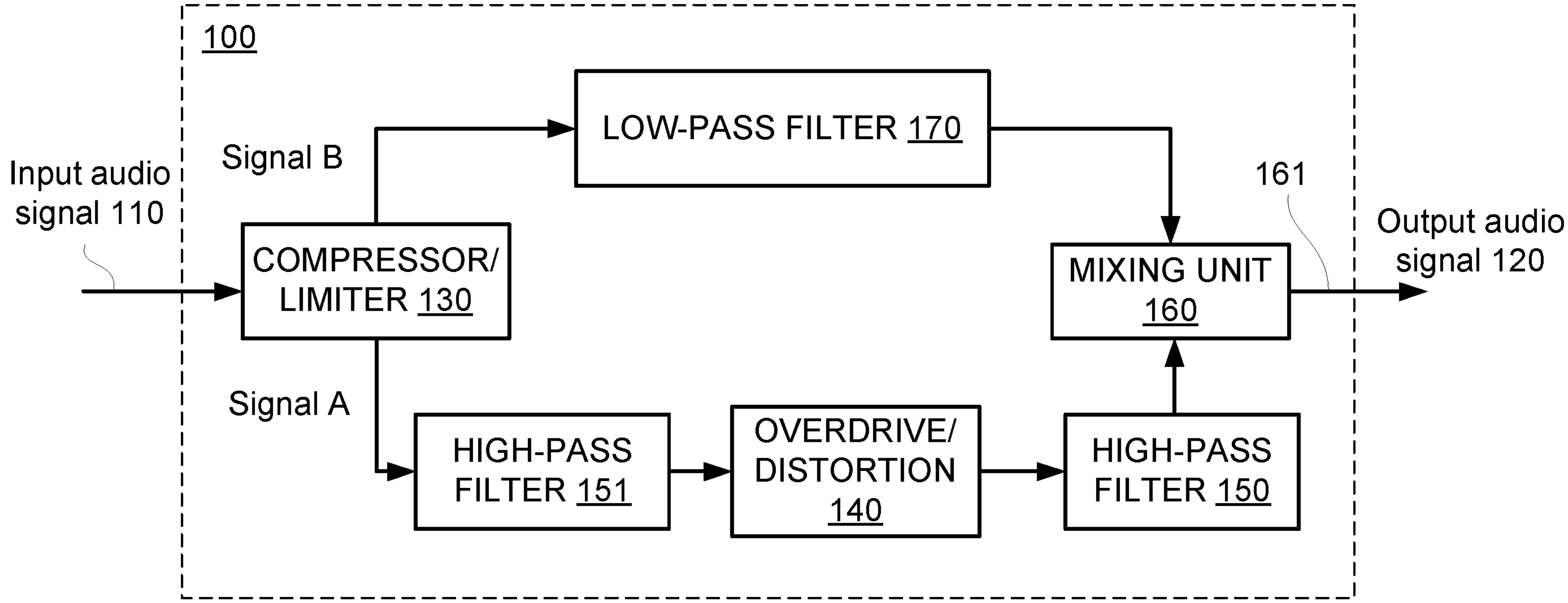

100
Input audio signal 110
Signal B
Signal A
COMPRESSOR/ LIMITER 130
LOW-PASS FILTER 170
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
HIGH-PASS FILTER 150
MIXING UNIT 160
161
Output audio signal 120

Fig. 3.

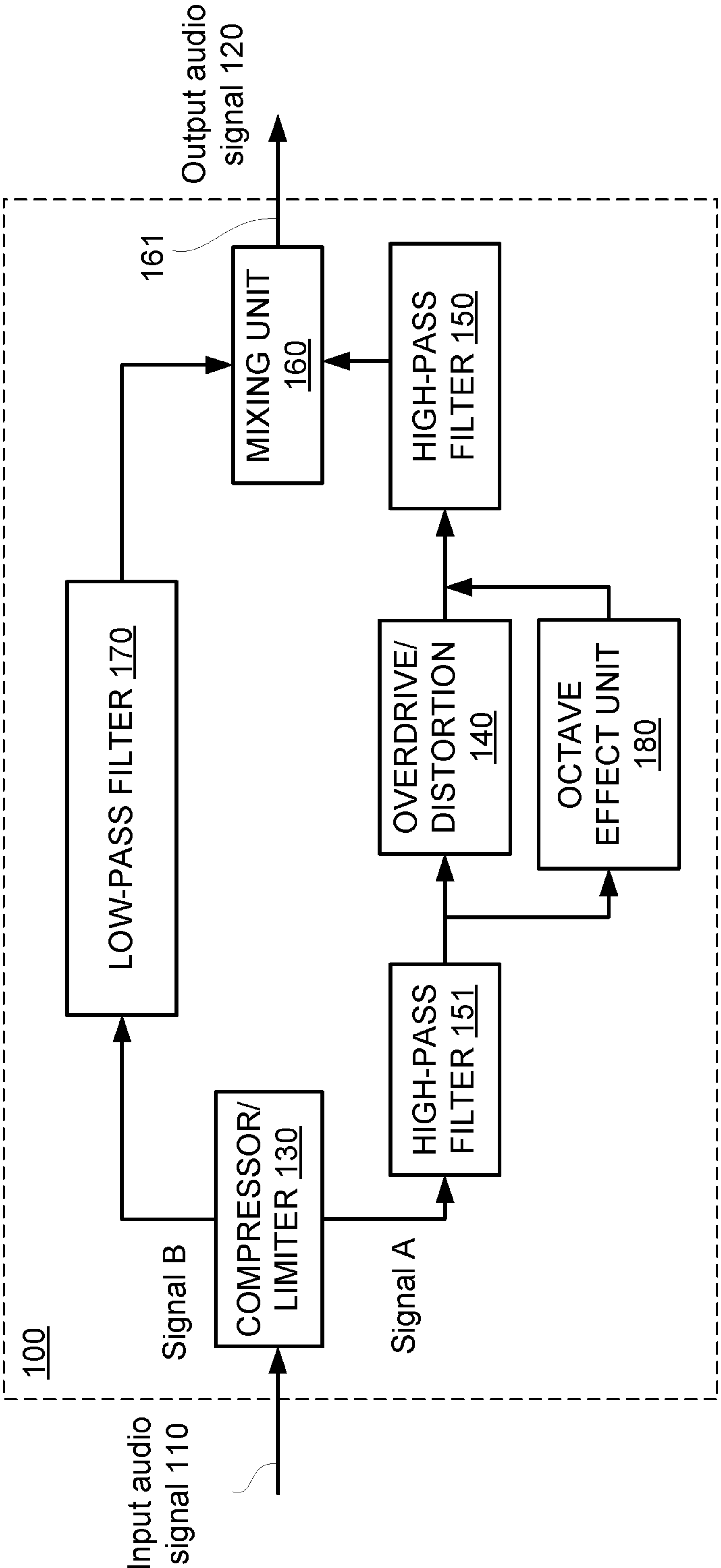
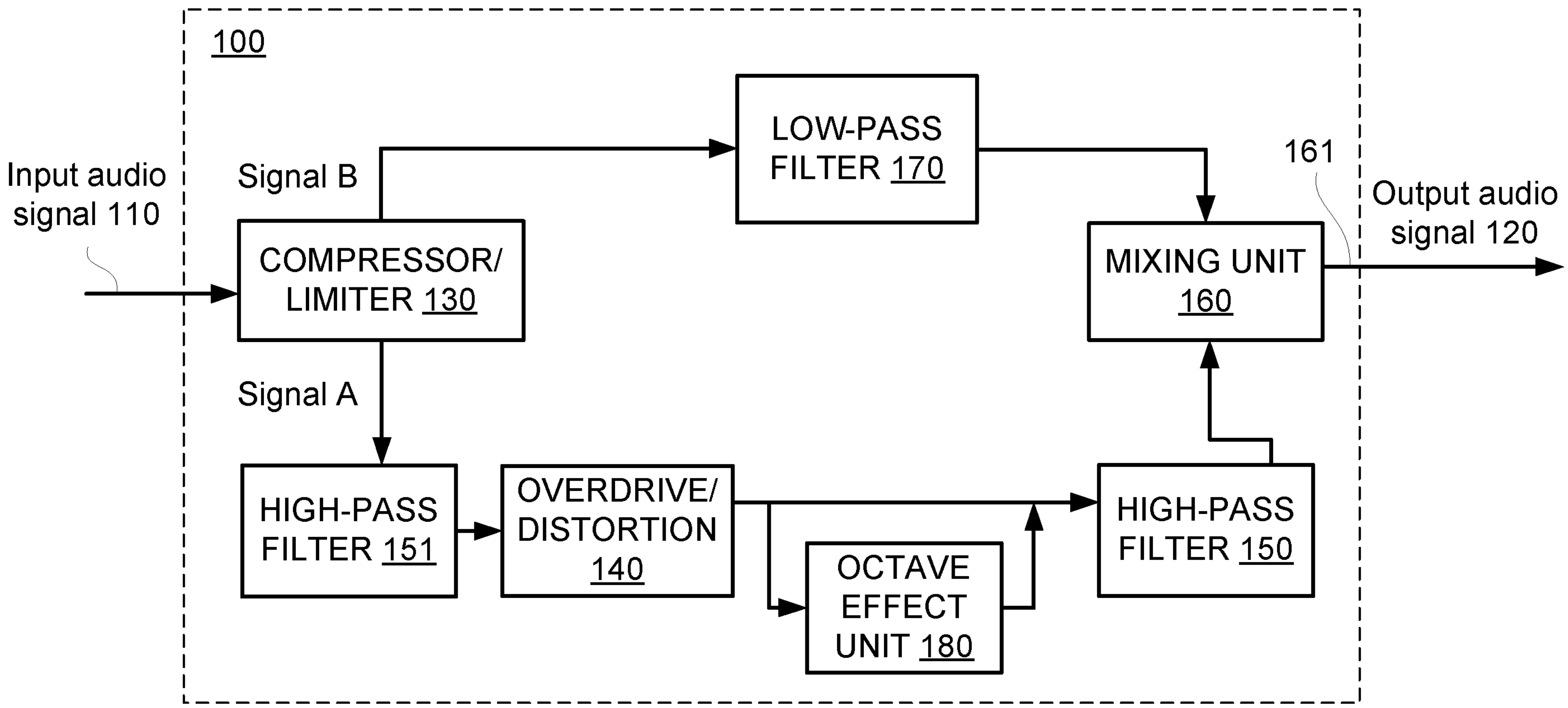
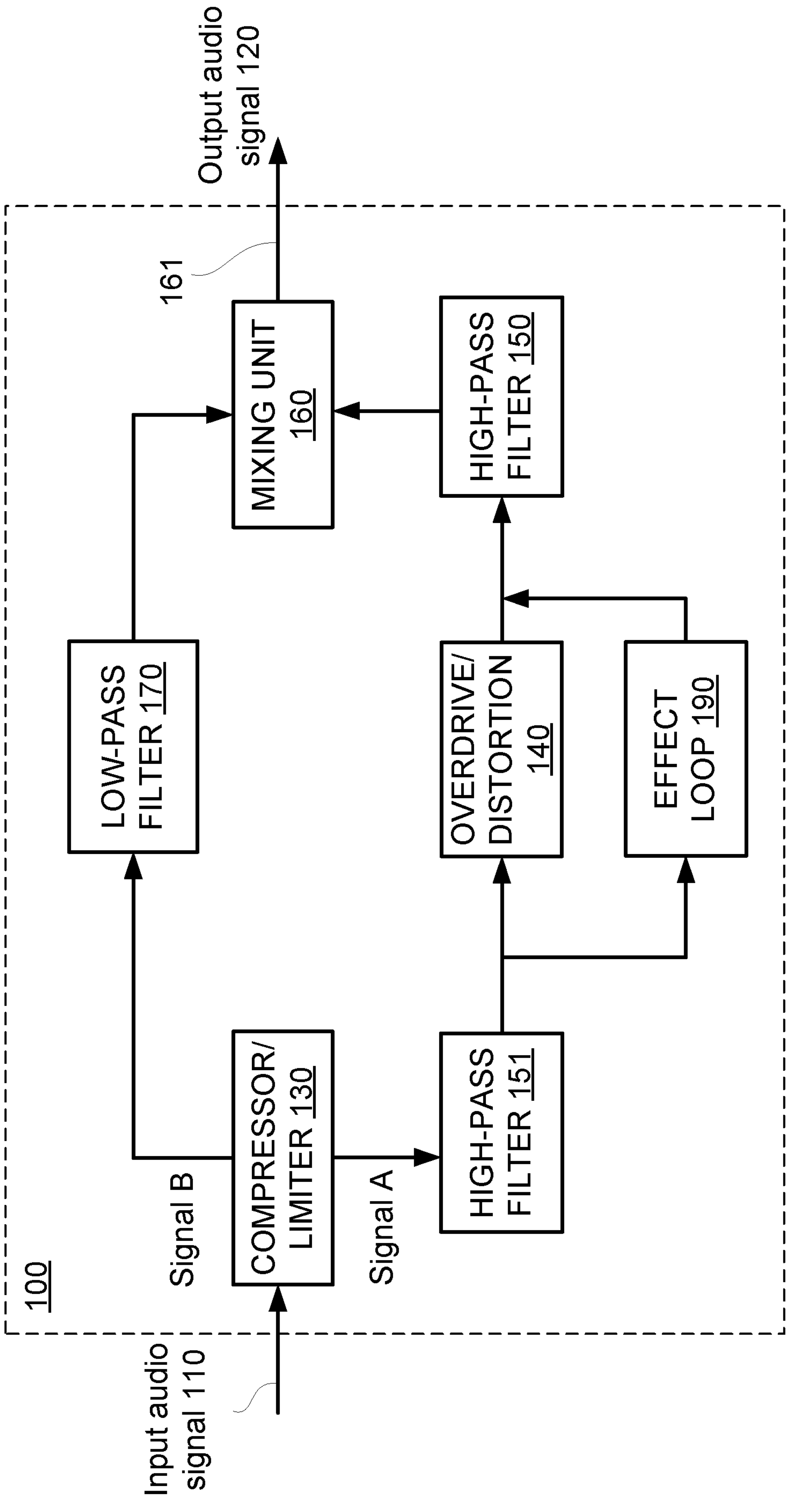

Output audio signal 120
161
100
MIXING UNIT 160
HIGH-PASS FILTER 150
LOW-PASS FILTER 170
OVERDRIVE/ DISTORTION 140
OCTAVE EFFECT UNIT 180
HIGH-PASS FILTER 151
COMPRESSOR/ LIMITER 130
Signal B
Signal A
Input audio signal 110

Fig. 4a.



100
Input audio signal 110
Signal B
Signal A
COMPRESSOR/ LIMITER 130
LOW-PASS FILTER 170
161
Output audio signal 120
MIXING UNIT 160
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
OCTAVE EFFECT UNIT 180
HIGH-PASS FILTER 150

Fig. 4b.



Output audio signal 120
161
100
Input audio signal 110
Signal B
Signal A
COMPRESSOR/ LIMITER 130
LOW-PASS FILTER 170
MIXING UNIT 160
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
EFFECT LOOP 190
HIGH-PASS FILTER 150

Fig. 5a.

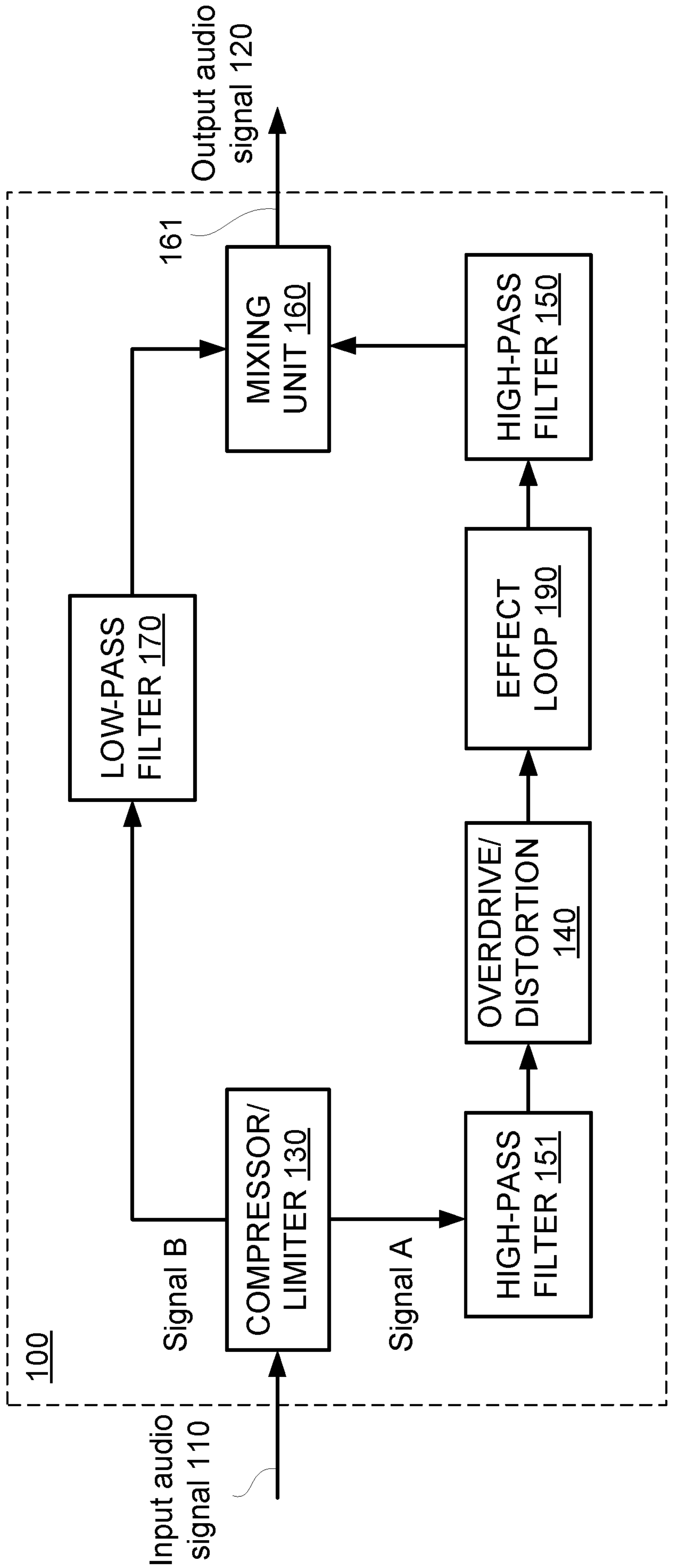

Output audio
signal 120
161
MIXING
UNIT 160
HIGH-PASS
FILTER 150
LOW-PASS
FILTER 170
EFFECT
LOOP 190
OVERDRIVE/
DISTORTION
140
COMPRESSOR/
LIMITER 130
HIGH-PASS
FILTER 151
Signal B
Signal A
100
Input audio
signal 110

Fig. 5b.

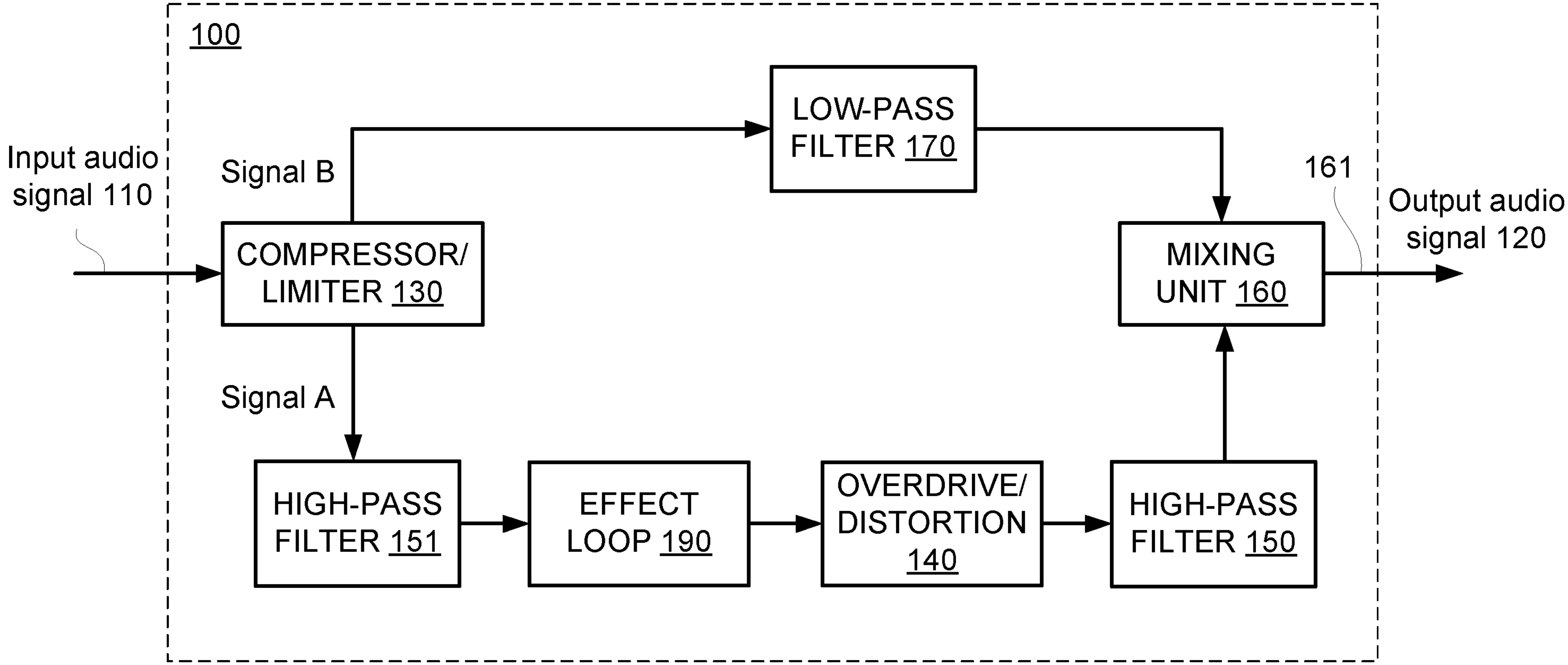

100
Input audio
signal 110
Signal B
COMPRESSOR/
LIMITER 130
Signal A
LOW-PASS
FILTER 170
MIXING
UNIT 160
161
Output audio
signal 120
HIGH-PASS
FILTER 151
EFFECT
LOOP 190
OVERDRIVE/
DISTORTION
140
HIGH-PASS
FILTER 150

Fig. 5c.

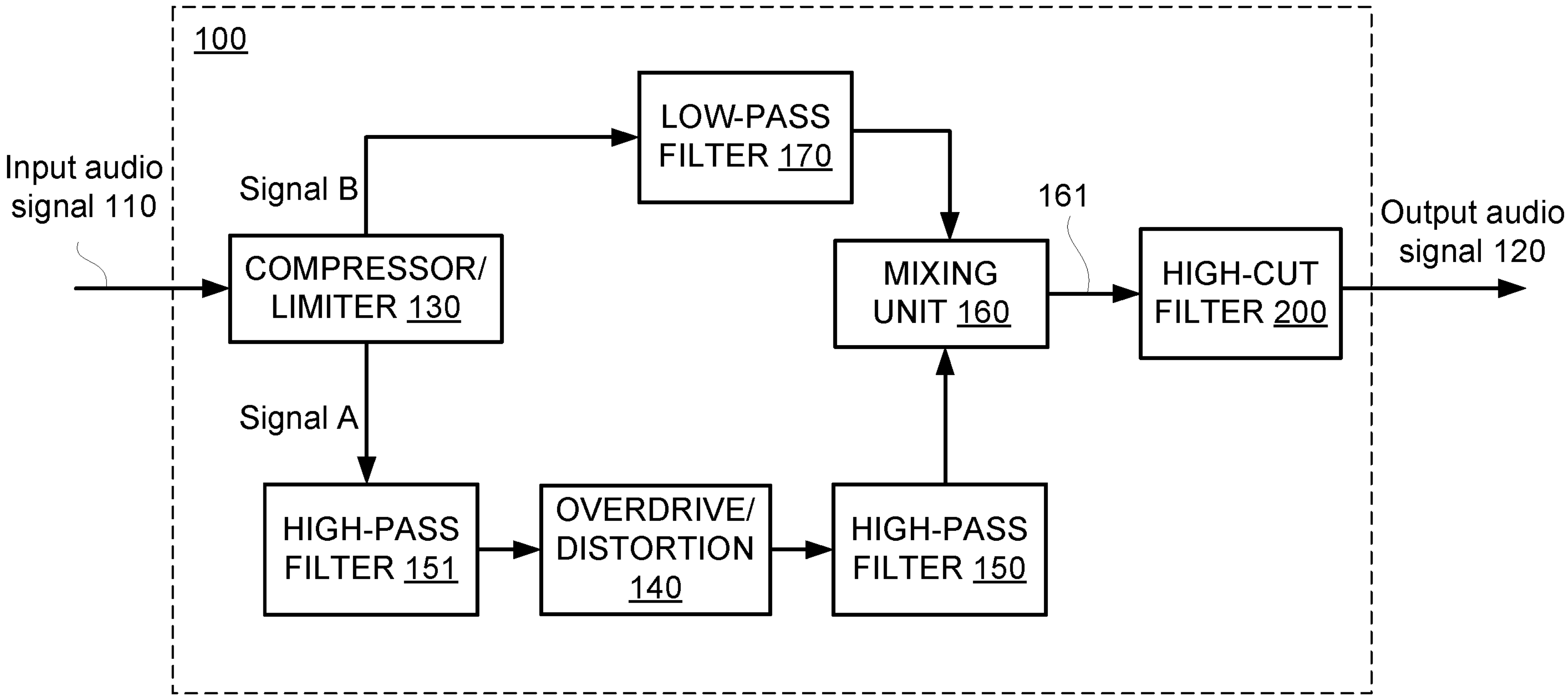

100
Input audio signal 110
Signal B
Signal A
COMPRESSOR/ LIMITER 130
LOW-PASS FILTER 170
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
HIGH-PASS FILTER 150
MIXING UNIT 160
161
HIGH-CUT FILTER 200
Output audio signal 120

Fig. 6.

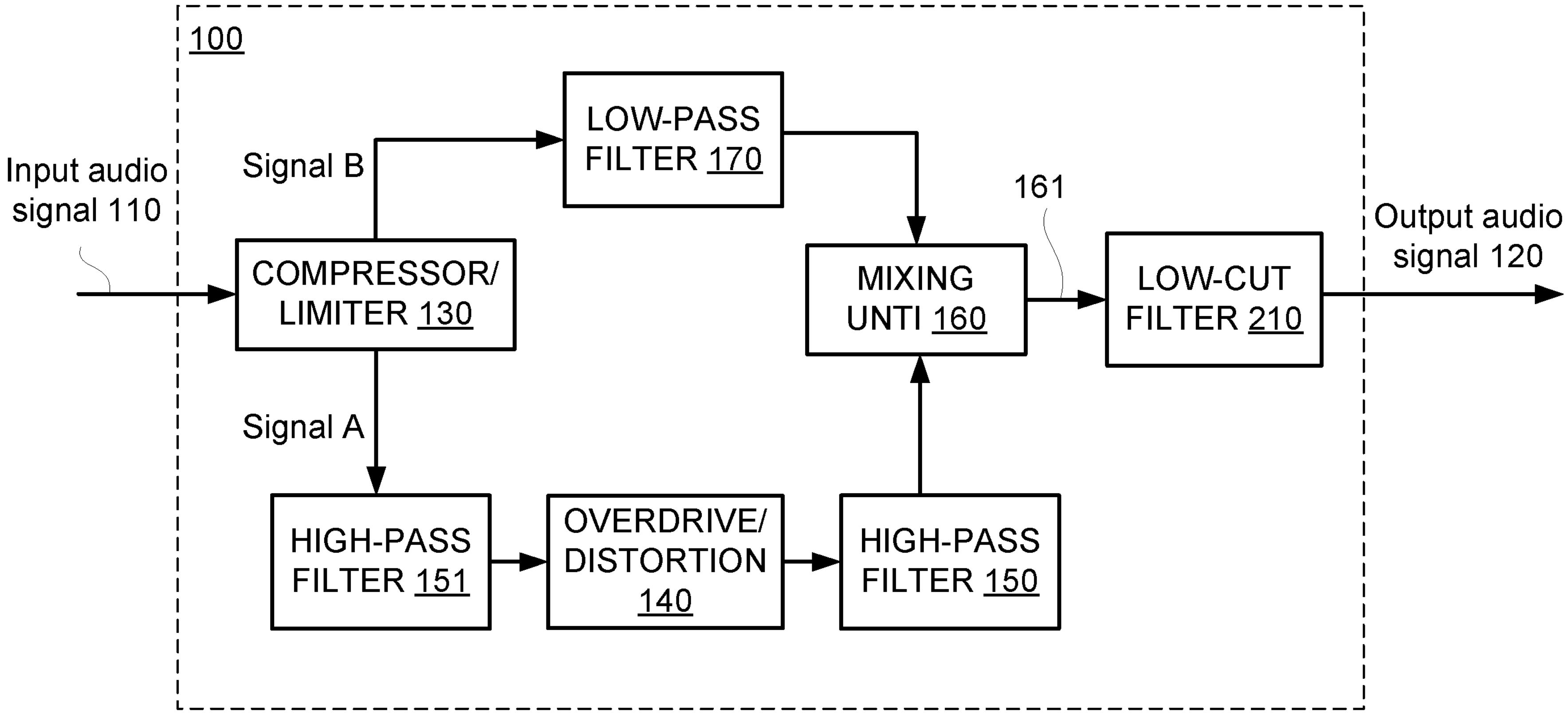

100
Input audio
signal 110
Signal B
Signal A
COMPRESSOR/
LIMITER 130
LOW-PASS
FILTER 170
HIGH-PASS
FILTER 151
OVERDRIVE/
DISTORTION
140
HIGH-PASS
FILTER 150
MIXING
UNTI 160
161
LOW-CUT
FILTER 210
Output audio
signal 120

Fig. 7.

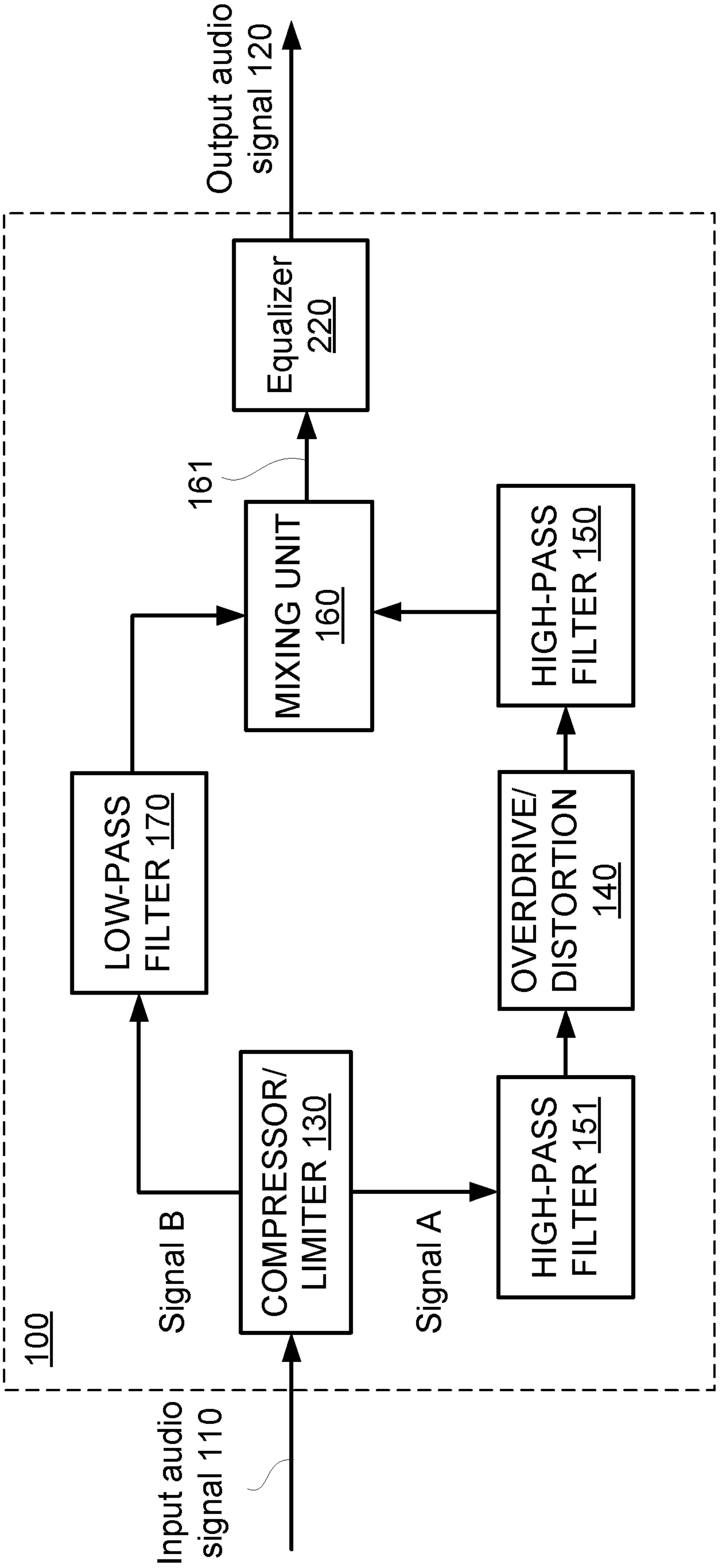

Output audio signal 120
Equalizer 220
161
MIXING UNIT 160
HIGH-PASS FILTER 150
LOW-PASS FILTER 170
OVERDRIVE/ DISTORTION 140
COMPRESSOR/ LIMITER 130
HIGH-PASS FILTER 151
Signal B
Signal A
100
Input audio signal 110

Fig. 8.

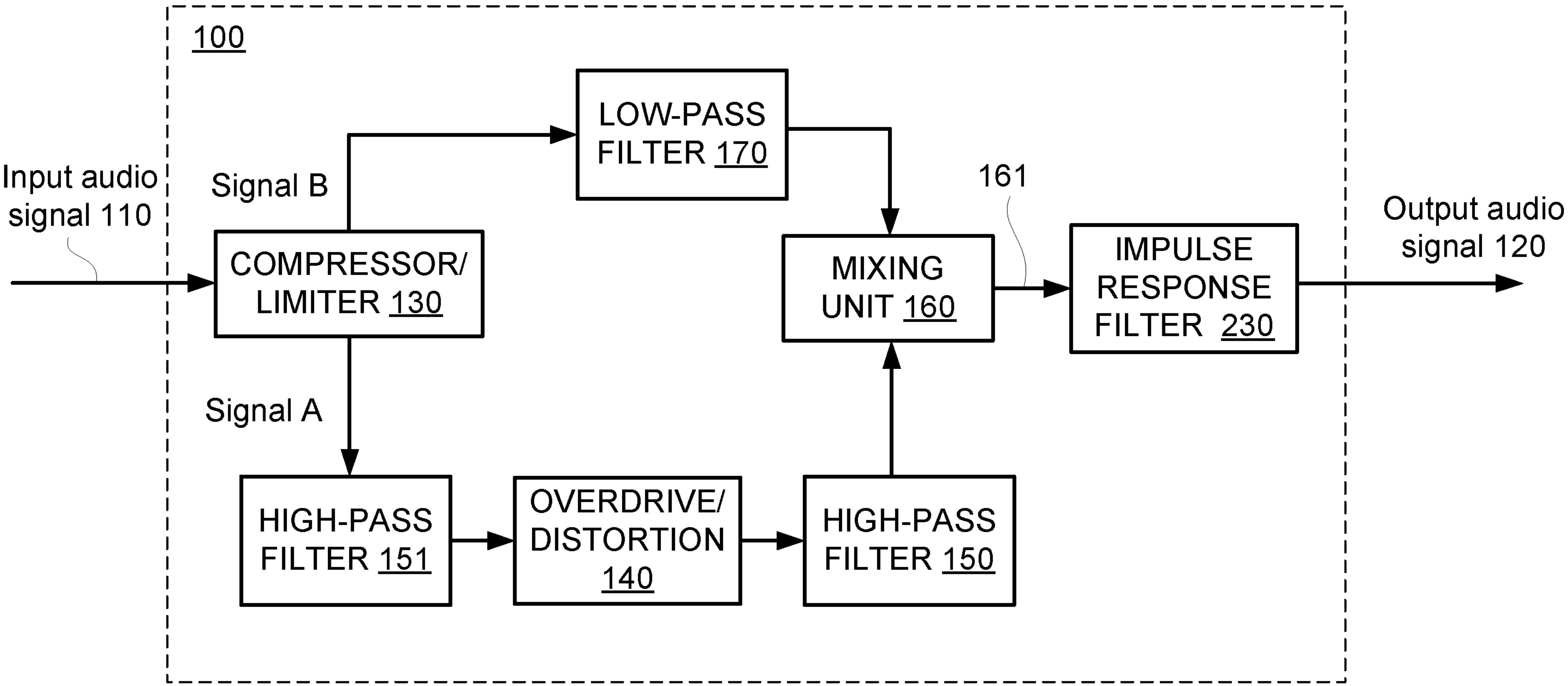

100
Input audio signal 110
Signal B
COMPRESSOR/ LIMITER 130
Signal A
LOW-PASS FILTER 170
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
HIGH-PASS FILTER 150
MIXING UNIT 160
161
IMPULSE RESPONSE FILTER 230
Output audio signal 120

Fig. 9.

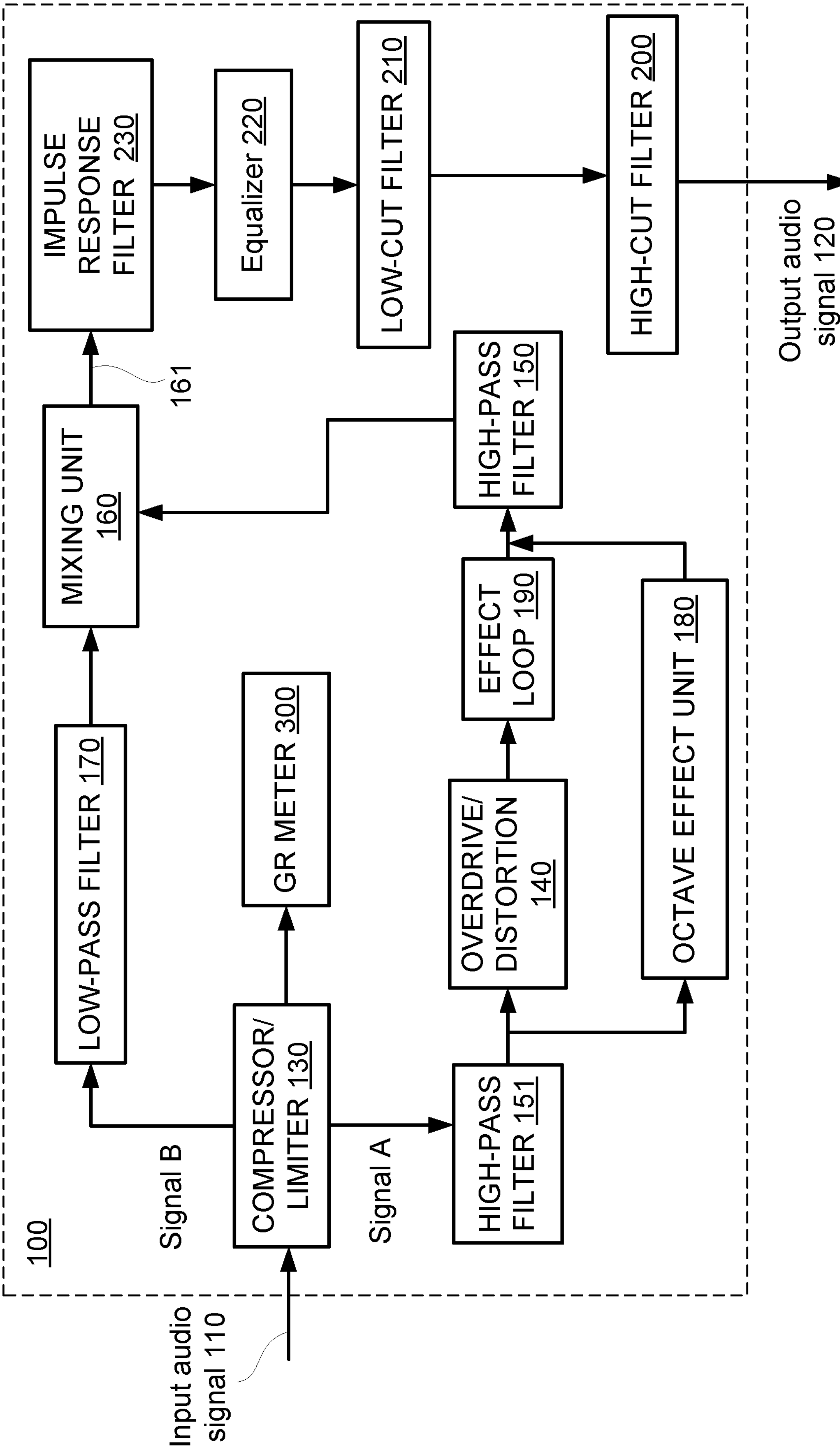

100
Input audio signal 110
COMPRESSOR/ LIMITER 130
Signal B
Signal A
GR METER 300
LOW-PASS FILTER 170
HIGH-PASS FILTER 151
OVERDRIVE/ DISTORTION 140
OCTAVE EFFECT UNIT 180
EFFECT LOOP 190
HIGH-PASS FILTER 150
MIXING UNIT 160
161
IMPULSE RESPONSE FILTER 230
Equalizer 220
LOW-CUT FILTER 210
HIGH-CUT FILTER 200
Output audio signal 120

Fig. 10.