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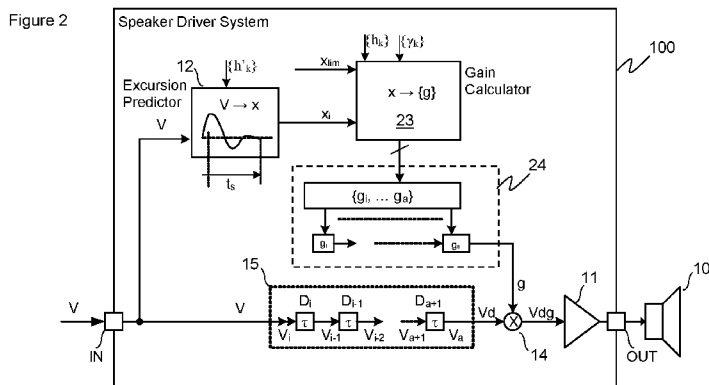
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(54) Title: LOUDSPEAKER PROTECTION

(57) Abstract: This application relates to methods and apparatus for loudspeaker protection. A loudspeaker protection system (100) receives a digital audio signal comprising a plurality of samples at an input node (IN). A delay block (15) delays the digital audio signal and a gain block (14) applies a controlled gain to the delayed digital audio signal. An excursion predictor (12) is configured to receive a version of the audio signal from the signal path upstream of the delay block and determine a predicted excursion for a loudspeaker based on the audio signal. A gain controller (23) controls a gain setting (g) of the gain block in response to the predicted excursion and a first loudspeaker impulse response model and a predetermined excursion limit. The gain controller (23) is configured to determine at least one gain setting $\{g_a, g_b\}$ to be applied to a set of samples $\{V_a, \dots, V_b\}$ of the audio signal delayed by the delay block where the set of samples of the audio signal corresponds to a time-span which is less than an impulse response settling time of the loudspeaker.

Loudspeaker Protection

The field of representative embodiments of this disclosure relates to methods, apparatuses, and/or implementations concerning and/or relating to protecting
5 loudspeakers and/or other similar output transducers, and more particularly to controlling the drive signal supplied to a loudspeaker so as to avoid and/or limit excessive diaphragm excursion.

A number of different products include audio circuitry, such as an audio amplifier,
10 together with one or more loudspeakers and/or connections for driving one or more loudspeakers of an integrated apparatus such as a mobile phone, i.e. handset, and/or peripheral apparatus such as a headset, e.g. earbuds, headphones, hearing aids and Bluetooth™ devices. In some instances the loudspeaker(s) chosen will be robust enough and sized sufficiently to handle the maximum power level at which the amplifier
15 could drive signals continuously into it, even under the worst case environmental conditions, for instance maximum supply voltage, maximum ambient temperature etc. However having robust enough loudspeakers is not always economical, and for portable devices such as mobile phones or tablets and headsets and the like the desire is typically to make the speaker as small and light as possible. This can potentially
20 lead to the audio drive circuitry overloading the loudspeaker. One particular problem is mechanical damage due to excessive and/or prolonged displacement or excursion of the speaker mechanism.

It is known to provide circuitry to estimate the displacement of a speaker over time from
25 the voltage applied to the speaker using a plant model, i.e. a model of how the speaker reacts, and to reduce the applied signal if over-excursion is predicted. **Figure 1** illustrates a speaker driver system for driving loudspeaker 10 via output amplifier circuitry 11. The system receives, at an input node IN, an input signal V representing the output voltage V_{out} to be driven into the loudspeaker 10. The input signal V is also
30 applied to an excursion prediction block 12 which uses a plant model to provide an estimate x of the speaker coil excursion. Gain calculator block 13 then compares this predicted excursion x with some predefined excursion or displacement limit x_{lim} . If the estimated excursion x exceeds the limit x_{lim} then a reduced gain value is calculated, for example equal to x_{lim}/x_i , and a corresponding reduced gain is applied to the input

signal, e.g. by a gain adjustment element 14 that is upstream of the loudspeaker in the signal path, so as to attenuate the input signal and hence reduce the speaker excursion to less than the limit x_{lim} .

5 The loudspeaker 10 may have substantial time constants associated with its mechanical inertia and the inertia of the air volume to be driven. This may be expressed in terms of the settling time of its impulse response which may extend over several milliseconds. There will thus be a delay between when the input signal V is received by the speaker 10 and when the full extent of its effect on the voice coil excursion is experienced and thus recognised by the excursion prediction block 12. Thus the input signal V is passed via a delay block 15 to apply a delay d in the signal path between the signal input node IN and the gain adjustment element 14 so that the reduced gain may be applied to a delayed version V_d of the input signal stream early enough, i.e. to the portion of the signal stream V_d that actually would have produced the over-excursion if it had been allowed to be applied to the loudspeaker 10.

This delay d may be acceptable in some applications, but in some applications such as telephony, e.g. wireless communication, especially in speakerphone type operation, such a delay d might cause problems and therefore be disadvantageous.

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Embodiments of the present invention provide methods and/or apparatus for speaker protection that at least mitigate at least some of the above mentioned disadvantages.

The description below sets forth example embodiments according to this disclosure.

25 Further example embodiments and implementations will be apparent to those having ordinary skill in the art. Further, those having ordinary skill in the art will recognize that various equivalent techniques may be applied in lieu of, or in conjunction with, the embodiments discussed below, and all such equivalents should be deemed as being encompassed by the present disclosure.

30

Thus according to an aspect of the present invention there is provided a loudspeaker protection system comprising:

an input node for receiving a digital audio signal comprising a plurality of samples;

- a delay block for delaying the digital audio signal;
- a gain block for applying a controlled gain to the delayed digital audio signal output from the delay block;
- an excursion predictor configured to receive a version of the audio signal from
- 5 the signal path upstream of the delay block and determine a predicted excursion for a loudspeaker based on the audio signal; and
- a gain controller for controlling a gain setting of the gain block in response to the predicted excursion and a first loudspeaker impulse response model and a predetermined excursion limit,
- 10 wherein the gain controller is configured to determine at least one gain setting to be applied to a set of samples of the audio signal delayed by the delay block; and
- wherein said set of samples of the audio signal correspond to a time-span which is less than an impulse response settling time of the loudspeaker.
- 15 The set of samples may comprise a sample of the audio signal delayed by the delay block that corresponds to the most recent sample of the audio signal on which the relevant predicted excursion is based and a predetermined number of preceding samples.
- 20 In some embodiments the excursion predictor is configured to determine a predicted excursion in response to each new sample of the audio signal. In this case, in response to each predicted excursion the gain controller may be configured to determine the at least one gain setting to be applied to a respective set of samples.
- 25 The gain controller may be configured to determine a set of gain settings, the set of gain settings comprising a respective gain setting for each of the samples in said set.
- In some embodiments each of the gain settings of said set of gain settings have the same value as one another.
- 30 In some embodiments the gain controller may be configured to determine the set of gain settings according to a gain profile. The set of gain settings may be based on a common gain factor applied to all gain settings and a gain profile value that is predetermined for the respective gain setting in the set. In some examples the gain

profile may correspond to a linear ramp in gain, although other gain profiles may also be used. The gain controller may be configured to determine the common gain factor for the set of gain settings based on the predicted excursion and an excursion limit. In some instances the gain profile may be variable based on a gain profile control setting.

5

The first loudspeaker impulse response model may comprise a first series of impulse response coefficients. The impulse response coefficients of the first series may be variable based on a coefficient setting.

- 10 The excursion predictor may be configured to determine the predicted excursion based on the audio signal and a second loudspeaker impulse response model. The second loudspeaker impulse response model may comprise a second series of impulse response coefficients. The impulse response coefficients of the second series may be variable based on a coefficient setting. In some embodiments the second loudspeaker
- 15 impulse response model may be the same as the first loudspeaker impulse response model, however different models may be used in some embodiments.

- The delay block delays the audio signal for a delay time which may be less than the impulse response settling time of the loudspeaker. For example the delay time may be
- 20 in the range of 80% to 15% of the impulse response settling time of the loudspeaker. In some embodiments the delay time of the delay block is less than or equal to 2ms, or less than or equal to 1ms or less than or equal to 500 μ s.

- The set of samples of the audio signal may correspond to a time-span which in the
- 25 range of 80% to 15% of the impulse response settling time of the loudspeaker. In some embodiments the time-span of the set of samples may be less than or equal to 2ms, or less than or equal to 1ms or less than or equal to 500 μ s.

- The impulse response settling time of the loudspeaker may be seen as the duration of
- 30 the first two lobes of impulse response. The impulse response settling time of the loudspeaker may be seen as the time taken for an impulse response envelope to decay to 5% of its peak value. The impulse response settling time of the loudspeaker may be seen as the time taken for a step response to settle to 1%.

In some embodiments the gain controller may determine a first set of gain settings for a first set of samples of the audio signal delayed by the delay block based on a first predicted excursion and subsequently determines at least a second subsequent set of gain settings for a second set of samples of the audio signal delayed by the delay block based on a subsequent predicted excursion. The second set of samples may partly overlap with the first set of samples. In such a case the gain controller may be configured to use, for any sample in both the first and second sets, the lowest respective gain setting of the first and second sets of gain settings that correspond to that sample.

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In some embodiments the gain controller is further configured to determine said at least one gain setting for the set of samples based on a constraint on velocity of the loudspeaker coil.

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In some embodiments the gain controller is configured to determine the at least one gain setting for a set of samples in a frequency band that corresponds to a subset of the frequency range of the audio signal.

20

The loudspeaker protection system as described in any of the variants above may be implemented as an integrated circuit.

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In some embodiments an apparatus comprising a loudspeaker protection system as described in any of the variants above may be arranged downstream of another, i.e. a first, excursion limiter.

Aspects also relate to an apparatus comprising a loudspeaker protection system as described in any of the variants above.

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The apparatus may further comprise a digital-to-analogue converter downstream of the gain block and/or an amplifier downstream of the gain block.

The apparatus may further comprise a loudspeaker configured to be driven by a drive signal wherein the drive signal is derived from the output of the gain block.

The apparatus may be at least one of: a portable device; a battery powered device; a wearable device; a computing device; a communications device; a gaming device; a mobile telephone; a personal media player; a laptop, tablet or notebook computing device.

5

In another aspect there is provided a loudspeaker excursion limiter comprising:

an input for receiving digital input signal comprising a stream of samples;

an output for outputting an audio signal for a loudspeaker;

10 a signal path from said input to said output comprising a delay block and a gain block, wherein the gain block is downstream of the delay block and applies a gain controlled by a gain calculation block; and

an excursion predictor block configured to estimate loudspeaker excursion based on the digital input signal and a first loudspeaker model;

wherein:

15 said gain calculation block determines a set of gains to be applied to a sequential set of delayed samples based on the currently estimated loudspeaker excursion, a predefined excursion limit and a loudspeaker impulse response model; and the time-span of said sequential set of delayed samples is less than a loudspeaker impulse response settling time.

20

In a further aspect there is provided a loudspeaker protection system comprising:

a signal path for an audio signal between an input node and an output node, said audio signal path comprising a delay block for delaying the audio signal and a gain block for applying a controlled gain to the audio signal output from the delay block;

25 an excursion predictor configured to receive a version of the audio signal from the signal path upstream of the delay block and determine a predicted excursion for a loudspeaker based on the audio signal; and

a gain controller for controlling a gain setting of the gain block in response to the predicted excursion and a loudspeaker impulse response model,

30 wherein the gain controller is configured to determine at least one gain setting to be applied to a segment of the audio signal delayed by the delay block so as to limit the predicted excursion to an excursion limit;

wherein said segment of the audio signal has a duration which is less than a first duration, the first duration corresponding to an impulse response settling time of the loudspeaker.

- 5 Aspects also relate to a system for limiting the excursion of an output transducer comprising:

an excursion predictor for determining a predicted excursion of an output transducer to an audio signal received at an input node; and

- 10 a gain controller for controlling the gain applied to the audio signal by a gain block in response to the predicted excursion and an excursion limit;

wherein the propagation delay in the audio signal path between the input node and the gain block is less than an impulse response settling time of the output transducer.

- 15 The propagation delay may for instance be within the range of 80% to 15% of the impulse response settling time of the output transducer.

Aspects also relate to a system for limiting the excursion of an output transducer comprising:

- 20 an excursion predictor for determining a predicted excursion of an output transducer to an audio signal received at an input node; and

a gain controller for controlling the gain applied to the audio signal by a gain block in response to the predicted excursion and an excursion limit; and

- 25 a delay block for applying a delay in the audio signal path between the input node and the gain block, wherein said delay is less than 1ms.

Also provided is a system for limiting the excursion of an output transducer comprising:

an excursion predictor for determining a predicted excursion of an output transducer to an audio signal received at an input node; and

- 30 a gain controller for controlling the gain applied to the audio signal by a gain block in response to the predicted excursion and based on an excursion limit and an impulse response model of the output transducer.

Embodiments of the invention will now be described, by way of example only, with reference to the accompanying drawings, of which:

Figure 1 illustrates a speaker driver system providing excursion limiting;

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Figure 2 illustrates a speaker driver system comprising an excursion limiter according to an embodiment;

Figure 3 illustrates waveforms illustrating the operation of a speaker limiter as

10 illustrated in Figure 2;

Figure 4 illustrates one example of a gain profile;

Figure 5 illustrates plots showing how excursion error and gain settings vary with the time over which the gain change is applied for a system according an embodiment of the invention that implements a step change in gain;

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Figure 6 illustrates plots similar to those of figure 5 for an embodiment that implements a ramped gain profile;

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Figure 7 illustrates how excursion error varies with the time over which the gain change is applied for a conventional excursion protection system;

Figure 8 illustrates a comparison of the excursion error versus time duration of figures 5 to 7 and how it compares with impulse response time;

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Figure 9 illustrates an apparatus having an excursion limiter according to an embodiment preceded by another excursion limiter;

Figure 10 illustrates plots of the operation of an apparatus such as illustrated in figure 9;

30

Figure 11 illustrates a flow chart for calculating a gain adjustment factor according to one example; and

Figure 12 illustrates an example of an apparatus having a speaker protection system.

5

Description

Figure 2 illustrates a speaker driver system comprising an embodiment of a speaker protection system or speaker protection block 100 that provides excursion limiting protection for a loudspeaker.

Note that as used herein the term 'block' shall be used to refer to a functional unit or module which may be implemented at least partly by dedicated hardware components such as custom defined circuitry and/or at least partly be implemented by one or more software processors or appropriate code running on a suitable general purpose processor or the like. A block may itself comprise other blocks or functional units.

As in Figure 1, this speaker protection block 100 receives an input audio signal V at an input terminal or input node IN and provides an output signal Vout at an output terminal or output node OUT for forwarding to a loudspeaker 10, e.g. via a driver amplifier 11. The input signal V may be a stream of digital samples at some suitable sample rate say 48ks/s. The digital samples may be of some suitable resolution as required to give suitable dynamic range in terms of maximum range and quantisation noise. The driver amplifier block 11 may comprise digital-to-analogue conversion circuitry so as to apply an analogue signal to the loudspeaker 10. The input signal V may be derived from some upstream analogue signal via an analogue-to-digital convertor.

The input signal V is also applied to an excursion prediction block 12 which uses an electro-mechanical plant model to provide an estimate x of the speaker coil excursion. The plant model may comprise coefficients $\{h'_k\}$ corresponding to an electro-mechanical model of the speaker, for example coefficients of the speaker impulse response or step response or of a digital filter of some topology. This model may be characterised by a settling time t_s , which may for example be defined as: the duration of some number of lobes, say three, of the speaker impulse response; the time taken

for the envelope of the impulse response to decay to say 5% of peak magnitude; the time taken for the envelope of the step response to decay to say 1% of the peak value; a multiple of the time constant of the lowest frequency pole of the response; or some other similar metric of the speaker electro-mechanical response corresponding to the
 5 duration of any significant effect or any combination of the aforementioned.

Gain calculator block 23 then accepts this estimated speaker excursion x and a predetermined value of the excursion limit x_{lim} and generates an appropriate gain value g to be applied to a delayed version V_d of the input signal V by gain element 14,
 10 which is located in the signal path before the loudspeaker 10, so as to attenuate the delayed input signal V_d by a factor g and hence reduce the speaker excursion to less than the provided maximum displacement value x_{lim} .

However in contrast to the arrangement illustrated in Figure 1, rather than the gain
 15 values being set equal to the current value of the ratio of x/x_{lim} , possibly with some attack or decay or hold dynamics being used to ensure the gain reduction spans the whole settling time t_s of the speaker response, in this embodiment a set of gain values $\{g_a \dots g_i\}$ are calculated together, to be applied to a respective set of successive samples of the delayed audio signal V_d , where the set of samples to which the set of
 20 gains are applied corresponds to a total time-span much shorter than the settling time t_s . As discussed in more detail below, merely setting the gain to the current value of the ratio of x/x_{lim} will not be adequate to avoid over-excursion, therefore an alternative set of calculations needs to be performed.

25 In some embodiments the gain values in a given set $\{g_a \dots g_i\}$ will all be equal to one another in order to maximise their total effect: in other embodiments or modes the gain values may follow some profile, for example a ramp or S-shape, in order to avoid any audio artefacts the may result from sudden gain changes. Such a profile may be determined by a set of coefficients $\{y_k\}$ that are input to the gain calculator 23. In some
 30 embodiments the gain calculation may use the impulse response of the speaker which may be defined by coefficients $\{h_k\}$ that are input to the gain calculator 23.

The respective set of gain values $\{g_a \dots g_i\}$ to be applied to the respective delayed input signal stream samples V_d include gains to be applied to past input signal samples, i.e.

those which have already been input to the excursion predictor block 12. Thus the signal V_{dg} in the main signal path via the gain block 14 to the loudspeaker 10 needs to be delayed prior to the gain block 14 and after an input IN to the excursion predictor block 15 is tapped off. Thus a delay block 15 is inserted in this path as shown.

5

The delay block 15 is illustrated as a series chain of delay elements, D_a...D_i, each delaying the signal by one sample period. In some embodiments the delay elements may be implemented using other means, for instance as an area of memory addressed by suitable read-write pointers, e.g. a circular buffer.

10

Figure 3 illustrates waveforms illustrating the operation of a speaker over excursion limiter block as illustrated in Figure 2.

Figure 3(a) illustrates an input signal waveform V as applied to the input node IN of the speaker driver system. Though drawn as a continuous curve, this waveform may comprise a set of discrete digital samples {V_k}.

Based on these voltage samples {V_k}, excursion prediction block 12 generates a corresponding sequence {x_k} of estimates over time of the excursion x of the speaker voice coil over time, as illustrated in Figure 3(c), again illustrated as a continuous curve, and in this case with only discrete excursion estimate samples {x_a...x_i} explicitly drawn for simplicity.

In this embodiment, the estimated excursion is calculated using the impulse response {h_k} of the loudspeaker. At a time point t_i defined by a time index i the predicted excursion x_i is calculated by the convolution equation:

$$x_i = \sum_{k=-\infty}^i V_k \cdot h_{i-k} \quad (1)$$

This equation can be viewed as doing a point-by-point multiplication of the voltage and time-reversed impulse response and subsequently summing these terms. Figure 3(b) thus illustrates the impulse response in a time reversed fashion for ease.

30

In practice only a finite number of coefficients, say m, of the impulse response will be provided and used to calculate the predicted excursion. Ideally this number m should

be enough to span all significant terms of the impulse response, i.e. to span the full settling time t_{sf} of the speaker electro-mechanical response. Figure 3(a) illustrates a set of discrete samples V_i to V_{i-m} which may be used in conjunction with the m coefficients $h_0 \dots h_{m-1}$ of the impulse response illustrated in Figure 3(b) to generate the
 5 single sample x_i of the estimated excursion x illustrated in Figure 3(c).

In some embodiments, the excursion may be calculated by an excursion predictor block 12 which employs other types of calculation rather than the impulse response, for example some infinite-impulse response digital filter, with coefficients to determine the
 10 poles and zeros of the response.

In this example the input voltage waveform and modelled speaker characteristics are such that the excursion prediction block 12 first predicts an over-excursion relative to the defined excursion limit x_{lim} at a time t_i . The “ i ” subscript is the time index when
 15 this first predicted over-excursion event occurs.

As illustrated in Figure 2, the system comprises a delay or buffer 15, for instance a sequence of delay stages between the signal input node IN and the gain block 14. By running the pre-buffered audio samples V through the excursion predictor 12 the
 20 system will predict, before the delayed version V_d of the input signal V arrives at the gain block 14 or loudspeaker 10, that the excursion will exceed its limit x_{lim} . Thus the buffer 15 will allow the system a time window in which the gain calculator 23 may calculate an appropriate gain reduction and in which the gain block 14 may apply such attenuation to the delayed audio samples V_d so that the actual excursion of the
 25 loudspeaker will remain within its limit x_{lim} .

The gain calculator block 23 thus calculates a series of gain settings, g_k , i.e. a gain profile such as illustrated in plot (d) of Figure 3, which will reduce the predicted excursion at time t_i to its prescribed limit. Using Equation (1) we can mathematically
 30 state this constraint as:

$$x_{lim} = \sum_{k=-\infty}^i g_k \cdot V_k \cdot h_{i-k} \quad (2)$$

If the predicted excursions with and without gain adjustment are calculated by equations (2) and (1) respectively, and if g_k is assumed to have a constant value G for all k , then x_{lim} will be equal to $G \cdot x_i$, so a gain reduction by a factor equal to x_{lim}/x_i will be adequate to maintain x_i equal to x_{lim} . In practice, only a finite number, m , of points
 5 may be used to calculate x_i , but provided m is large enough to span all significant values of the speaker impulse response, then a gain of x_{lim}/x_i will still be adequate to maintain the excursion within the predetermined limit. In other words were the excursion x_i predicted (without any gain adjustment) in response to a new sample to exceed the predetermined limit, then by applying a gain reduction equal to x_{lim}/x_i to the
 10 m preceding samples the predicted excursion after gain limiting would remain within acceptable limits, provided that the set of preceding m samples is large enough.

However, in order to be able to apply the correct gain to input sample V_{i-m-1} which occurred m sample periods earlier, the delay introduced by delay block 15 has to be at
 15 least m sample periods. Such a delay may be unacceptable in some applications. In some embodiments therefore a gain value or set of gain values may be calculated that may be applied to a set of successive samples that span a time period which is shorter than the settling time of the loudspeaker but which are sufficient to maintain the excursion within acceptable limits.

20 Thus assuming that only the last n samples V_{i-n-1} to V_i are still buffered in delay block 15 and available to be subjected to gain reduction, and that the nominal gain of the gain block 14 is unity, then effectively in equation (2) the gains g_k for $k=-\infty \dots i-n-1$ should be taken as unity.

25 The gain calculator may thus determine a sequence of gains $\{g_a \dots g_i\}$ that are applied to the samples V_a to V_i output from the delay block 15 so as to maintain the loudspeaker excursion within acceptable limits.

30 In some embodiments, the sequence of gains $\{g_a \dots g_i\}$ applied may be described in terms of a calculated gain adjustment factor G and a predefined gain profile sequence $\{\gamma_k\}$ such that the sequence of applied gains is given by:

$$g_k = 1 + G \cdot \gamma_k \quad (3)$$

Where γ_k may be non-zero over a range of k from i to a .

For example, as illustrated in Figure 4, γ_k may be given by

$$\gamma_k = \begin{cases} 0 & \text{if } k < a \\ -\frac{k-a}{i-a} & \text{otherwise} \end{cases} \quad (4)$$

In this example γ_k is zero for $k = a$ and -1 for $k = i$ and ramps down linearly between
 5 these two values. The applied gain $\{g_k\}$ is thus unity until $k = a$, and then ramps down to
 1-G by $k = i$.

With this decomposition there is now only one unknown parameter, G , and it is linearly
 related to the applied gains, g_k . This allows us to solve for G . Substituting (3) into (2)
 10 yields:

$$x_{lim} = \sum_{k=-\infty}^i (1 + G \cdot \gamma_k) \cdot V_k \cdot h_{i-k} \quad (5)$$

However this expression may be simplified considering (4):

$$x_{lim} = \sum_{k=-\infty}^i V_k \cdot h_{i-k} + G \sum_{k=a}^i \gamma_k \cdot V_k \cdot h_{i-k} \quad (6)$$

Finally the term on the left may be recognized as the previously calculated over
 excursion value x_i from equation (1). Substituting and solving for the gain yields:

$$G = \frac{x_{lim} - x_i}{\sum_{k=a}^i (V_k \cdot \gamma_k) \cdot h_{i-k}} \quad (7)$$

15 It can be seen that a value G is calculated as a general gain factor for a gain profile to
 be applied to a finite set of the n preceding samples, where the set of samples may
 have any desired length and may in particular be shorter than the settling response
 time of the loudspeaker.

20 The solution given in (7) may also be extended to gain profiles other than the ramp
 shown in Figure 4. If γ_k is set to -1, or some other constant value, for $a < k < i$ then a
 simple step change in gain will be implemented. An S-curve, or any other profile may
 be used provided that it may be expressed in the form of equation (3). The only
 difference is that a new γ_k will need to be defined instead of that defined in equation
 25 (4).

Note that for the case of a step change (y_k set to -1) and for an infinitely long buffer, i.e. $a = i - n - 1 = -\infty$, the denominator of equation (7) becomes equal to x_i , so G becomes $(x_{lim}/x_i) - 1$ and the applied gain becomes x_{lim}/x_i . Provided that n is large enough to encompass all the substantially non-zero terms of the speaker impulse response, the denominator will still be substantially equal to x_i and the applied gain necessary to avoid over-excursion will be equal to x_{lim}/x_i . Conversely applying the steady-state gain factor x_{lim}/x_i as in the scheme illustrated in figure 1 will be adequate to avoid over-excursion.

However for smaller values of n , i.e. for smaller ranges from a to i over which gain adjustment is applied, the denominator will generally be smaller and hence G necessary to avoid over-excursion will be larger in magnitude, i.e. the gain adjustment will need to be more severe. Just applying a gain of x_{lim}/x_i will not be sufficient. In other words, applying just this steady-stage gain x_{lim}/x_i will not be adequate unless the signal can be corrected over a sequence of signal values at least as long as the significantly non-zero portion of the speaker impulse response, i.e. the speaker settling time t_s .

In order to be able to apply the gain adjustment to such a long sequence of signal values, this sequence of values will need to be stored in the buffer 15, so the signal delay d through the buffer 15 must be at least as long as the speaker response settling time t_s . Providing such a long buffer 15 may be undesirable in terms of the extra hardware and memory requirements required. Also in some applications a signal delay d as prolonged as t_s is undesirable, for instance in telephony, particularly in speakerphone applications. The small extra computational effort required to perform the above gain factor calculations may thus be worthwhile.

Figure 3(d) thus illustrates the calculated gain settings determined as described above.

Figure 3(e) illustrates the delayed and gain-adjusted waveform V_{out} actually applied to the speaker 10 compared to that of the non-gain adjusted non delayed input signal V waveform of Figure 3(a) and Figure 3(f) illustrates the corresponding actual excursion waveform with and without the gain adjustment. The former may be seen to have successfully limited excursion so that x_i is reduced to x_{lim} .

The buffer 15 is thus sufficient to apply a delay sufficient for the n sample V_a to V_i . In addition to allow for signal propagation and processing of the signals in the excursions predictor and gain calculator path there may be some additional delay to allow for
5 synchronisation. However the delay implemented by the buffer may be shorter than the settling time of the loudspeaker impulse response.

It will be appreciated that the set of gains calculated $\{g_a \dots g_i\}$ should be applied to samples V_a to V_i output from the buffer or delay block 15. The gain settings may
10 therefore be stored in a gain buffer 24 with the gain buffer being designed with respect to the delay block 15 and the processing delays inherent in the excursion predictor 12 and gain calculator 23 so that the gain settings are applied to the appropriate set of samples. The set of samples of the delayed audio signal comprises the sample corresponding to the most recent sample on which the excursion prediction was made,
15 i.e. V_i , and a plurality of preceding samples, i.e. the samples through to V_a . The gain buffer 24 may therefore be arranged such that gain setting g_a is synchronised with the sample V_a at the gain element 14. The gain buffer, or similar synchronisation control, may form part of the gain controller in practice and is illustrated separately in figure 2 for clarity.

20

Figures 5(a), 5(b), 5(c) illustrate the advantageous reduction in signal delay made possible by employing a speaker driver system similar to that illustrated in Figure 2, with the gain profile set to give a step change in applied gain. To generate these plots, the buffer delay d was varied from near zero to about 4ms and the maximum over-
25 excursion and calculated gain adjustment factor G plotted versus the buffer delay. The multiple overlaid curves are the results of applying different input waveforms each with a similar potential over-excursion.

Figure 5(a) illustrates the resulting over-excursion, which can be seen to be zero
30 except for buffer delays smaller than about 300 μ s. In other words as long as the buffer had a delay of at least 300 μ s then calculating and applying the gain reductions as described above was sufficient to prevent over-excursion. This minimum buffer length compares favourably with the settling time of the time-reversed speaker impulse response shown in Figure 5(b). Figure 5(c) shows the gain adjustment factor G . For

long buffer delays this factor converges to about 0.25, corresponding to the gain of $(1 - 0.25) = 0.75$ that would be applied by a conventional system. For shorter buffer delays, since the gain will be applied to fewer samples, a more drastic gain reduction is required, until at about $300\mu\text{s}$ the calculated gain adjustment factor G becomes unity, corresponding to an actual applied gain of zero. Further reduction in the buffer delay causes the system to fail to prevent significant over-excursion.

Figure 6 illustrates similar plots to those illustrated in figure 5 but this time shows the results of using a linear ramp gain profile (rather than step change used for the plots of figure 5). Since the full gain adjustment G is not applied over the full range of delay, a larger value of G is required, and the system starts to fail at a longer delay of about $600\mu\text{s}$.

Figure 7 illustrates the results of a conventional system, where a gain of x_{lim}/x_i is applied. In other words figure 7 illustrates the effect of varying buffer size on a conventional system where the gain setting is controlled to the x_{lim}/x_i .

It can be seen that the conventional system requires a much longer minimum delay of around 3ms , of the order of the settling time of the speaker impulse response. For buffer delays of less than about 3ms the system is not capable of preventing over-excursion.

Figure 8 summarises the results illustrated in Figures 5 to 7, illustrating the advantages of the fast limiter, i.e. a system as per the embodiments disclosed herein that can operate with a reduced buffer delay, over the simple limiter. Similar to Figures 5 to 7 the top plot shows over excursion error versus the time-span over which the gain profile is applied. The fast limiter results using both step and ramp profiles are shown. The simple limiter results are overlaid for comparison. The bottom plot (b) is the system impulse response. Note that with both step and ramp profiles the fast limiter is able to maintain zero over-excursion error over a larger range of start times than the simple limiter. More specifically the gain profile of the fast limiter can be initiated much closer to the over excursion event. Therefore the fast limiter can be advantageously used with a much shorter delay buffer than the simple limiter.

In embodiments therefore the gain controller may thus determine at least one gain setting to be applied to a set of successive samples of the delayed audio signal where the time-span of the set of samples, i.e. the duration of the set, is less than a first duration which corresponds to an impulse response settling time of the loudspeaker. The settling response impulse time or the first duration may be seen as the duration of the first two lobes of impulse response, or the time taken for an impulse response envelope to decay to 5% of its peak value. The settling time/first duration could be the time taken for a step response to settle to 1%.

The delay time of the delay block 15, i.e. the length of the buffer may thus be less than the first duration/ impulse response settling time of the loudspeaker.

The time-span of the set of samples may for instance be in the range of 80% to 15% (inclusive) of the impulse response settling time of the loudspeaker. For instance the time-span may be not greater than 50% of the impulse response settling time of the loudspeaker.

Likewise the delay time for the loudspeaker may be in the range of 80% to 15% (inclusive) of the impulse response settling time of the loudspeaker and in some embodiments may be equal to or lower than 50% of the impulse response settling time of the loudspeaker.

In some embodiments, for the type of loudspeakers envisaged for the results of figure 8, this may mean that the time delay of the delay block and/or time-span of the set of samples may be less than or equal to 2ms, or less than or equal to 1ms or less than or equal to 500µs. For other types of loudspeaker however different time delays may be appropriate.

The advantages of the above described fast limiter include the following:

- it allows more flexibility in the choice of the delay of the system (i.e. the size of the buffer 15). As shown in Figure 8, the chosen gain profile of the fast limiter can be applied later than what the simple limiter would allow to avoid over-exursion events. This flexibility can be important when latency constraints are

stringent and the decay of the impulse response of the speaker exceeds such requirements;

- the gain profile used by the fast limiter can be chosen from a variety of different types since it has been clearly shown that the concept of fast limiting to prevent over excursion can be extended to gain profiles that are different from the step profile. To reduce distortion and allow smoother gain transitions when over-excursion is predicted, more appropriate profiles can be used such as linear ramps or S-shaped curves for example.

While the gain profile {gk} curves may define an attack time of the fast limiter, for example in terms of the ramp or s-shape duration, in some embodiments the gain calculator may continue to: hold the final reduced gain value for some extra number of samples; apply only a gradual gain recovery back to unity; or also implement and define a hold time and/or release times or release characteristic. These gains may be overridden should a later signal cause an over-excursion to be predicted. The profile and duration of the gain release can be selected to minimise any audible artefacts for example in the presence of repetitive or persistent overload conditions.

It should be noted that the fast limiter as disclosed herein, e.g. the speaker protection block of figure 2, may be used in combination with other types of excursion limiters. For example as illustrated in **figure 9** a first excursion limiter 901 may be used before a fast excursion limiter 902, which may be an excursion limiter according to the embodiments described above, e.g. speaker protection block 100. The fast excursion limiter 902 is thus a second limiter located downstream of the first limiter 901. The first limiter 901 could be designed to meet some specific system requirements (e.g. latency, MIPS, memory...) or some performance requirements (e.g. loudness, frequency response...). However in order to meet such requirements, the first limiter 901 may not be able to guarantee complete excursion protection (e.g. reduced latency could cause excursion overshoots similar to those shown in Figure 8 for the simple limiter), but it may still attenuate excursion effectively enough to cover most of the potential over-excursion events. In this case, the fast limiter 902 may be applied after limiter 901 to catch any residual overshoots that passed through the first limiting stage. With limited additional delay and minimal audible impact (given by the chosen curve applied to the gain profile), limiter 902 can guarantee full protection and reduce only the remaining excursion peaks that could damage the speaker.

Figure 10 shows an example of the combined behaviour of a first limiter followed by a fast limiter such as illustrated in figure 9. Figure 10(a) illustrates the excursion with and without the excursion limiting. Figure 10(b) illustrates the gain adjustment applied by the first limiter 901, which in this case is a multi-band limiter that applies excursion limiting to different frequency bands of the audio signal and so the gain adjustments for each band are illustrated. Figure 10(c) illustrates the gain adjustment applied by the fast limiter i.e. limiter 902.

- It can be seen that the multi-band limiter is often active in attenuating excursion peaks and effectively reduces the great majority of excursion peaks bringing them down to the safe level. The fast limiter (bottom plot) intervenes relatively infrequently, and with small gain adjustments, to further reduce the samples that would cause excessive excursion. In various embodiments comprising a plurality of cascaded limiters design choices may be made to trade-off the relative reliance on each limiter.

The above description describes how a potential over-excursion at a time index i may be prevented by applying a gain reduction to a set of n preceding signal samples. Despite this gain reduction, it may still be the case that the next signal sample may be such that a potential over-excursion may occur at the next time point. In such a case a similar calculation and gain adjustment may be applied to the respective set of n immediately preceding samples, i.e. to a set of samples displaced by one sample period in time, and any overlapping gain values simply over-written, or preferably the minimum value applicable to each gain coefficient may be used. Over-excursion protection will still be effective at each time point.

As illustrated in **Figure 11**, in some embodiments, the excursion calculation at each time point may be adjusted to take account of the gain reductions already prescribed. In other words, for each time point the excursion x_i is initially calculated using a set of values for gain $\{1 + g_k\}$ calculated at the previous time point, possibly including some hold time or decay dynamics applied to the gain values. If this newly calculated excursion for the current time point is less than x_{lim} , then this set of gains is retained. If not then a new value of G is calculated, along with corresponding revised gain values

$\{1 + G \cdot \gamma_k\}$ and the excursion is recalculated. This loop may be iterated until a satisfactory solution is found, as illustrated in the flow diagram of Figure 11.

In further embodiments, the gain control calculations may be modified to control both the position and the velocity of the speaker coil. The added velocity constraint may increase the robustness of the algorithm. With only a position constraint, it is possible that the excursion immediately before or after the constrained point may exceed the threshold. However, with an added velocity constraint, the excursion limiter may locally control the slope, i.e. rate of change, of the excursion verses time. For example, consider that a gain profile is chosen to force the excursion of a given point to be at the excursion limit. If the slope is simultaneously forced to be positive, points immediately before this point must have a smaller excursion, and therefore remain in the limit. If instead the slope is forced to be zero, points immediately before and after this point will remain at the limit. In this way a velocity constraint may be used to increase robustness.

The fast limiter as described in any of the variants above may be extended to include a velocity constraint. To accomplish this it will be necessary to use at least two gain profiles, g_k^1 and g_k^2 , along with the corresponding sequences γ_k^1 and γ_k^2 as defined in (3) and (4). Two gain profiles are sufficient for velocity but additional profiles could be used if desired, e.g. to extend to acceleration or higher derivatives.

The final gain profile, g_k , will be a weighted sum of these two gain profiles:

$$g_k = 1 + g_1 \cdot \gamma_k^1 + g_2 \cdot \gamma_k^2 \quad (8)$$

The scalar constants g_1 and g_2 are used to construct the final gain profile, g_k , in a manner similar to (3).

The methods used to derive (5) through (7) are re-applied, but now include a second set of linear equations for velocity. The result is that (7) is transformed into a vector equation:

$$\begin{bmatrix} x_i - x_{lim} \\ v_i - v_i^f \end{bmatrix} = \begin{bmatrix} \sum_{k=a}^i (V_k \cdot \gamma_k^1) \cdot h_{i-k} & \sum_{k=a}^i (V_k \cdot \gamma_k^2) \cdot h_{i-k} \\ \sum_{k=a}^i (V_k \cdot \gamma_k^1) \cdot r_{i-k} & \sum_{k=a}^i (V_k \cdot \gamma_k^2) \cdot r_{i-k} \end{bmatrix} \cdot \begin{bmatrix} g_1 \\ g_2 \end{bmatrix} \quad (9)$$

Where v_i is the initial velocity at time “i”, v_i^f is the final desired velocity at time “i”, and r_k is the impulse response of the velocity. All other variables in (9) are as they were defined in (7).

- 5 Note that (9) is a vector equation that must be simultaneously solved for g_1 and g_2 . The results are then substituted into (8) to derive the final gain sequence.

- In some embodiments, particularly for use without any preceding excursion limiter, it may be desirable to implement a multi-band form of the limiter, i.e. to be able to apply
 10 different gains to components of the signal which fall into different frequency bands. These frequency bands may be linearly spaced or logarithmically defined or defined in some other way for example mel frequency bands.

- The input signal V may be regarded as the superposition of many signals, say M
 15 signals:

$$v[n] = \sum_{m=0}^{M-1} v_m[n]$$

- Assuming linearity, the excursion prediction block may calculate an estimate for the total excursion at a given time point as a summation of the respective excursion
 20 components for each signal component:

$$\begin{aligned} x[n] &= \sum_{m=0}^{M-1} x_m[n] \\ &= \sum_{m=0}^{M-1} \sum_{i=0}^{\infty} v_m[n-i] * h[i] \end{aligned}$$

If $|x[n]| > X_{lim}$, then an over-excursion has been predicted and a gain reduction needs to be applied to a range of signal samples.

Similarly to the single-band case disclosed above, the condition for limiting the total excursion to x_{lim} may be expressed as:

$$x_{lim} = \sum_{m=0}^{M-1} \sum_{i=0}^{\infty} v_m[n-i] h[i] g_m[i]$$

where:

$$g_m[i] = (1 - G_m \gamma_m[i])$$

In some embodiments the ramping function may be the same for all bands. In other embodiments, each frequency band indexed m may have an independent ramping function or gain profile $\{g_m[i]\}$.

The equation may be simplified by substitution thus:

$$x_{lim} = x[n] - \sum_{m=0}^{M-1} G_m \sum_{i=0}^{K_m} v_m[n-i] h[i] \gamma_m[i]$$

Whereas in the single band case there was a single G to solve for to provide the gain reduction value, there are now m values to solve for. This is an underdetermined problem, and embodiments may add various other constraints to determine the relative gain adjustments to be defined by the gain calculation block in different bands.

In some embodiments the relative gains may be derived so as to minimise the magnitude sum $|G_m|$, i.e. to minimise the overall gain reduction, may be minimised. For example initially assume all $G_m = 0$ (no gain reduction) except for the band, say band index q , with the largest excursion, and solve for G_q explicitly using

$$x_{lim} - x[n] = -G_q \sum_{i=0}^{K_q} v_q[n-i] h[i] \gamma_q[i]$$

If $G_q < 1$ then this gain reduction is enough to limit the excursion. If $G_q > 1$ set G_q to unity to give the maximum attenuation possible in that frequency band q , and move on to perform a similar calculation for the band with the next largest excursion, and continue until the excursion limit has been realised. If $G_m < 0$ for any frequency band

set it to zero. Other optimisation criteria and methods [as per sister multiband excursion limit case] are possible.

5 In some embodiments, the signal samples may be processed in frames, for example of 16 samples per frame.

10 In some embodiments the gains may be calculated at every audio sample period, e.g. at 48ks/s. In other embodiments the gains may be calculated only once per frame of say 16 samples. In such cases the excursion value used to calculate the required gain reduction may be the maximum of the magnitudes of the estimated excursions for the current frame.

15 In some embodiments at least some of parameters of the speaker protection block may be configurable differently according to a user's use case. For example a longer signal delay may be permissible in music play-back use, with potentially high-quality source material, allowing less aggressive attack times for the gain modulation and less consequent manipulation of the original signal. On the other hand, for phone voice calls especially speakerphone-mode calls for example the delay may preferably be reduced to meet a lower latency budget. The parameters may therefore be configurable either
20 by a user or automatically. In some embodiments the parameters may be configurable in use, for instance a user could select certain parameters according to the their preference and/or defined sets of parameters could be selected based on the use, for instance an apps processor or the like could determine whether a media file is being played or whether voice call data is being relayed and configure speaker protection
25 accordingly.

The excursion prediction block 12 may calculate the excursion or displacement based on the coefficients of a stored impulse response. Alternatively some other model, for example a Thiele-Small Model may be used as the basis of calculations. In either case,
30 the coefficients of the displacement model may be fixed, maybe by initial design on the basis of some initial characterisation from pilot builds, or by a one-time calibration during manufacturing. The coefficients may be adapted during use on the basis of parameter estimates from the voltage and current waveforms in the load, and possibly

modified in use based on a detection of temperature of the voice coil or some other part of the speaker or host device or of the ambient temperature.

While the speaker mechanism and its excursion have been referred to as a voice coil,
5 this does not imply that the speaker need comprise a cylindrical coil – other geometries of coil and magnetic components may be employed. The speaker mechanism may employ other means that the electromagnetic effect of a coil acting on a permanent magnet. Indeed the general method is applicable to other transducers, for instance electrostatic, where a mechanical excursion requires being limited. The method may
10 also be applicable to limiting of other physical variables associated with a speaker. Although described in terms of the excursion of a loudspeaker cone, the invention is applicable to many types of audio output transducer. Applicable transducers may comprise various types of mechanical member of various geometries whose movement requires to be constrained to prevent damage or degradation over time, and may
15 include motor elements other than electromagnetic coils, for example piezo-electric drivers. Embodiments may be used to process other signals than conventional audio signals (20hz-20khz), for example ultrasonic signals, haptic stimulus signals, sub-audio signals and for the purposes of this disclosure such signals are considered as “audio signals”. In such cases the system may be adapted for use by substituting an
20 appropriate excursion response function.

In some embodiments the gain calculation circuitry may be temporarily disabled, for example the clocks may be removed from some or all of the relevant circuitry, should it be detected that the power dissipated in the coil or the predicted excursion is well
25 below some threshold. This may reduce computational overhead when not required and reduce power consumption.

The protection circuitry described above may be incorporated in audio amplifier circuitry in portable battery-powered devices such as mobile phones, tablets or lap-top
30 computers or suchlike. It may also be used in speakerphones, mains-powered music or PA amplifiers, audio amplifiers in automobiles and other transport apparatus.

Embodiments of the invention thus provide a speaker protection system that determines a speaker excursion based on an input signal tapped before a delay block

so as to control gain settings applied to the samples output from the delay block. In
embodiments the gain of a set of samples is controlled where the set of samples
includes a sample corresponding to the most recent sample used in the excursion
prediction and a plurality of preceding samples. The time-span of this set of samples
5 may be less than a first duration corresponding to the impulse response settling time of
the loudspeaker being driven. This means that the time delay of the delay block may
also be less than this duration. The gains may be calculated taking the impulse
response of the loudspeaker into account.

10 It will be noted that the calculated gain settings applied to just this set of samples is
sufficient to maintain the excursion of the loudspeaker within an acceptable limit. In
other words it is not necessary to control the gain of any samples preceding this set in
order to limit the currently predicted excursion. Thus in the event that a gain reduction
is required to limit a currently predicted excursion the gain reduction will only occur
15 within the set of samples. It will be appreciated that earlier samples will of course have
been subjected to gain control but the gain settings applied to such earlier samples will
have been determine based on earlier excursion estimates.

Embodiments also relate to a speaker protection system that determines a speaker
20 excursion based on an input signal tapped before a delay block so as to control gain
settings applied to the samples output from the delay block in which the gain of a set of
samples is controlled based on the impulse response of loudspeaker.

Figure 12 illustrates an embodiment of an apparatus 1200 incorporating the speaker
25 protection system according to the present invention.

The apparatus, for example a mobile phone or tablet, comprises the speaker protection
system 1201, which may be a system as described in any of the embodiments above.
The speaker protection system 1201 is arranged to receive an audio signal either from
30 an internal signal source 1202 or an external source.

An internal signal source 1202 may comprise a memory, e.g. a solid state memory,
arranged to store media with an audio component, such as music or video, for
playback via the speaker protection system 1201 and drive amplifier 1203 and finally at

least one speaker 1204, which may be an internal speaker of the device or may be part of a peripheral apparatus which is connected to the apparatus in use.

5 An external source may comprise a communications network such as those used for mobile and wireless communications wherein the apparatus has a receiver 1205 to receive a voice call or media file for playback via the speaker protection system 1201 and drive amplifier 1203 and finally at least one speaker 1204.

10 It will be appreciated by those skilled in the art that during a voice call there are timing and latency requirements that are beyond the control of the apparatus. Thus as mentioned previously any signal processing in the signal path between the receiver 1205 and speaker 1204 should not introduce any significant delay. As mentioned previously the embodiments described above provide speaker protection that introduces only relatively low delays which will typically be within any allowable latency
15 limits.

The speaker protection system 1201 may receive one or more feedback signals from the speaker for directly determining the displacement of the speaker itself and that are used to set the parameters of the displacement model within speaker protection
20 system. Such feedback signals may, for example comprise current and/or voltage signals.

In some embodiments the speaker protection system may receive one or more pre-programmed signals relating to operation of the speaker protection system. Such
25 signals may be based on data, settings or code that is internally stored, for example in a memory 1206. Such settings or data may be stored during manufacturing or during use in a calibration routine, e.g. as performed periodically or on power-up or reset. Such signals may be used to set at least some parameters of the speaker protection, such as attack or decay constant to be applied to gain changes etc., or data regarding
30 the displacement model within the speaker protection system.

It will of course be appreciated that various embodiments of speaker protection block as disclosed or various blocks or parts thereof may be co-integrated with other blocks

or parts thereof or with other functions of a host device on an integrated circuit such as a Smart Codec.

The skilled person will thus recognise that some aspects of the above-described apparatus and methods, for example the calculations performed by the processor may be embodied as processor control code, for example on a non-volatile carrier medium such as a disk, CD- or DVD-ROM, programmed memory such as read only memory (Firmware), or on a data carrier such as an optical or electrical signal carrier. For many applications embodiments of the invention will be implemented on a DSP (Digital Signal Processor), ASIC (Application Specific Integrated Circuit) or FPGA (Field Programmable Gate Array). Thus the code may comprise conventional program code or microcode or, for example code for setting up or controlling an ASIC or FPGA. The code may also comprise code for dynamically configuring re-configurable apparatus such as re-programmable logic gate arrays. Similarly the code may comprise code for a hardware description language such as Verilog[™] or VHDL (Very high speed integrated circuit Hardware Description Language). As the skilled person will appreciate, the code may be distributed between a plurality of coupled components in communication with one another. Where appropriate, the embodiments may also be implemented using code running on a field-(re)programmable analogue array or similar device in order to configure analogue hardware

Embodiments of the invention may be arranged as part of an audio processing circuit, for instance an audio circuit which may be provided in a host device. A circuit according to an embodiment of the present invention may be implemented as an integrated circuit. One or more loudspeakers may be connected to the integrated circuit in use.

The embodiments described above have been described with reference to reproduction of audio signals in the audio band, e.g. for speaker phone applications. The principles are equally applicable however to output transducers operating in other frequency bands that operate on similar principles. As used in this specification therefore the terms audio shall be taken to cover a range of frequencies including ultrasonic and/or infrasonic frequency bands. The term loudspeaker shall be interpreted accordingly to refer to any similar type of output transducer including

ultrasonic outputs transducers or haptic transducers for example. For the avoidance of doubt however in some embodiments the loudspeaker may particularly be a loudspeaker for reproducing audio signals that are in the normal audible band, e.g. in the range of say 20Hz to 20kHz.

5

Embodiments have also been described in terms of a digital input and digital gain adjustment. A digital implementation is preferred but the principles may be applied equally to at least some analogue elements. For instance a controlled analogue gain could be applied to a segment of an analogue audio and/or an analogue input signal could be received.

10

Embodiments may be implemented in a host device, especially a portable and/or battery powered host device such as a mobile telephone, an audio player, a video player, a PDA, a mobile computing platform such as a laptop computer or tablet and/or a games device for example. Embodiments of the invention may also be implemented wholly or partially in accessories attachable to a host device, for example in active speakers or headsets or the like.

15

It should be noted that the above-mentioned embodiments illustrate rather than limit the invention, and that those skilled in the art will be able to design many alternative embodiments without departing from the scope of the appended claims. The word "comprising" does not exclude the presence of elements or steps other than those listed in a claim, "a" or "an" does not exclude a plurality, and a single feature or other unit may fulfil the functions of several units recited in the claims. Any reference numerals or labels in the claims shall not be construed so as to limit their scope. Terms such as amplify or gain include possibly applying a scaling factor of less than unity to a signal.

20

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CLAIMS

1. A loudspeaker protection system comprising:
an input node for receiving a digital audio signal comprising a plurality of
5 samples;
a delay block for delaying the digital audio signal;
a gain block for applying a controlled gain to the delayed digital audio signal
output from the delay block;
an excursion predictor configured to receive a version of the audio signal from
10 the signal path upstream of the delay block and determine a predicted
excursion for a loudspeaker based on the audio signal; and
a gain controller for controlling a gain setting of the gain block in response to
the predicted excursion and a first loudspeaker impulse response model
and a predetermined excursion limit,
15 wherein the gain controller is configured to determine at least one gain setting
to be applied to a set of samples of the audio signal delayed by the
delay block; and
wherein said set of samples of the audio signal correspond to a time-span
which is less than an impulse response settling time of the loudspeaker.
20
2. A loudspeaker protection system as claimed in claim 1 wherein said set of
samples comprises a sample of the audio signal delayed by the delay block that
corresponds to the most recent sample of the audio signal on which the
relevant predicted excursion is based and a predetermined number of
25 preceding samples.
3. A loudspeaker protection system as claimed in claim 1 or claim 2 wherein the
excursion predictor is configured to determine a predicted excursion in
response to each new sample of the audio signal.
30
4. A loudspeaker protection system as claimed in claim 3 wherein, in response to
each predicted excursion the gain controller is configured to determine the at
least one gain setting to be applied to a respective set of samples.

5. A loudspeaker protection system as claimed in any preceding claim wherein the gain controller is configured to determine a set of gain settings, the set of gain settings comprising a respective gain setting for each of the samples in said set.
- 5 6. A loudspeaker protection system as claimed in claim 5 wherein each of the gain settings of said set of gain settings have the same value as one another.
7. A loudspeaker protection system as claimed in claim 5 wherein the gain controller is configured to determine the set of gain settings according to a gain profile.
10
8. A loudspeaker protection system as claimed in claim 7 wherein said set of gain settings is based on a common gain factor applied to all gain settings and a gain profile value that is predetermined for the respective gain setting in the set.
15
9. A loudspeaker protection system as claimed in claim 7 or claim 8 wherein said gain profile corresponds to a linear ramp in gain.
10. A loudspeaker protection system as claimed in claim 8 or claim 9 wherein the gain controller is configured to determine the common gain factor for the set of gain settings based on the predicted excursion and an excursion limit.
20
11. A loudspeaker protection system as claimed in any of claims 8 to 10 wherein the gain profile is variable based on a gain profile control setting.
25
12. A loudspeaker protection system as claimed in any preceding claim wherein the first loudspeaker impulse response model comprises a first series of impulse response coefficients.
- 30 13. A loudspeaker protection system as claimed in claim 12 wherein the impulse response coefficients of the first series are variable based on a coefficient setting.

14. A loudspeaker protection system as claimed in any preceding claim wherein the excursion predictor is configured to determine the predicted excursion based on the audio signal and a second loudspeaker impulse response model.
- 5 15. A loudspeaker protection system as claimed in any preceding claim wherein the second loudspeaker impulse response model comprises a second series of impulse response coefficients.
- 10 16. A loudspeaker protection system as claimed in claim 15 wherein the impulse response coefficients of the second series are variable based on a coefficient setting.
- 15 17. A loudspeaker protection system as claimed in claim 14 wherein the second loudspeaker impulse response model is the same as the first loudspeaker impulse response model.
- 20 18. A loudspeaker protection system as claimed in any preceding claim wherein the delay block delays the audio signal for a delay time which is less than the impulse response settling time of the loudspeaker.
- 25 19. A loudspeaker protection system as claimed in claim 14 wherein the delay time of the delay block is less than or equal to 2ms, or less than or equal to 1ms or less than or equal to 500 μ s.
- 30 20. A loudspeaker protection system as claimed in any of claims 1 to 18 wherein said set of samples of the audio signal correspond to a time-span which is less than or equal to 2ms, or less than or equal to 1ms or less than or equal to 500 μ s.
21. A loudspeaker protection system as claimed in any preceding claim wherein impulse response settling time of the loudspeaker is the duration of the first two lobes of impulse response.

22. A loudspeaker protection system as claimed in any of claims 1 to 20 wherein impulse response settling time of the loudspeaker is the time taken for an impulse response envelope to decay to 5% of its peak value.
- 5 23. A loudspeaker protection system as claimed in any of claims 1 to 20 wherein impulse response settling time of the loudspeaker is the time taken for a step response to settle to 1%.
- 10 24. A loudspeaker protection system as claimed in any preceding claim:
wherein the gain controller determines a first set of gain settings for a first set of samples of the audio signal delayed by the delay block based on a first predicted excursion and subsequently determines at least a second subsequent set of gain settings for a second set of samples of the audio signal delayed by the delay block based on a subsequent predicted excursion;
15 wherein the second set of samples partly overlaps with the first set of samples;
and
wherein the gain controller is configured to use, for any sample in both the first and second sets, the lowest respective gain setting of the first and
20 second sets of gain settings that correspond to that sample.
25. A loudspeaker protection system as claimed in any preceding claim wherein the gain controller is further configured to determine said at least one gain setting for the set of samples based on a constraint on velocity of the loudspeaker coil.
25
26. A loudspeaker protection system as claimed in any preceding claim wherein the gain controller is configured to determine said at least one gain setting for set of samples in a frequency band that corresponds to a subset of the frequency range of the audio signal.
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27. A loudspeaker protection system as claimed in any preceding claim implemented as an integrated circuit.

28. An apparatus comprising a loudspeaker protection system as claimed in any preceding claim arranged downstream of a first excursion limiter.
29. An apparatus comprising a loudspeaker protection system as claimed in any of
5 claims 1 to 27.
30. An apparatus as claimed in claim 27 further comprising a digital-to-analogue converter downstream of the gain block.
- 10 31. A loudspeaker protection system as claimed in claim 29 or claim 30 further comprising an amplifier downstream of the gain block.
32. An apparatus as claimed in any of claims 29 to 31 further comprising a
15 loudspeaker configured to be driven by a drive signal wherein the drive signal is derived from the output of the gain block.
33. An apparatus as claimed in any of claims 29 to 32 wherein the apparatus is at least one of: a portable device; a battery powered device; a wearable device; a
20 computing device; a communications device; a gaming device; a mobile telephone; a personal media player; a laptop, tablet or notebook computing device.
34. A loudspeaker excursion limiter comprising:
25 an input for receiving digital input signal comprising a stream of samples;
an output for outputting an audio signal for a loudspeaker;
a signal path from said input to said output comprising a delay block and a gain block, wherein the gain block is downstream of the delay block and applies a gain controlled by a gain calculation block; and
an excursion predictor block configured to estimate loudspeaker excursion
30 based on the digital input signal and a first loudspeaker model;
wherein:
said gain calculation block determines a set of gains to be applied to a sequential set of delayed samples based on the currently estimated

loudspeaker excursion, a predefined excursion limit and a loudspeaker impulse response model; and
the time-span of said sequential set of delayed samples is less than a loudspeaker impulse response settling time.

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35. A loudspeaker protection system comprising:
a signal path for an audio signal between an input node and an output node,
said audio signal path comprising a delay block for delaying the audio
signal and a gain block for applying a controlled gain to the audio signal
output from the delay block;
an excursion predictor configured to receive a version of the audio signal from
the signal path upstream of the delay block and determine a predicted
excursion for a loudspeaker based on the audio signal; and
a gain controller for controlling a gain setting of the gain block in response to
the predicted excursion and a loudspeaker impulse response model,
wherein the gain controller is configured to determine at least one gain setting
to be applied to a segment of the audio signal delayed by the delay
block so as to limit the predicted excursion to an excursion limit;
wherein said segment of the audio signal has a duration which is less than a
first duration, the first duration corresponding to an impulse response
settling time of the loudspeaker.

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36. A system for limiting the excursion of an output transducer comprising:
an excursion predictor for determining a predicted excursion of an output
transducer to an audio signal received at an input node; and
a gain controller for controlling the gain applied to the audio signal by a gain
block in response to the predicted excursion and an excursion limit;
wherein the propagation delay in the audio signal path between the input node
and the gain block is less than an impulse response settling time of the
output transducer.
37. A system for limiting the excursion of an output transducer comprising:
an excursion predictor for determining a predicted excursion of an output
transducer to an audio signal received at an input node; and

a gain controller for controlling the gain applied to the audio signal by a gain block in response to the predicted excursion and an excursion limit; and a delay block for applying a delay in the audio signal path between the input node and the gain block, wherein said delay is less than 1ms.

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38. A system for limiting the excursion of an output transducer comprising:

an excursion predictor for determining a predicted excursion of an output transducer to an audio signal received at an input node; and

a gain controller for controlling the gain applied to the audio signal by a gain

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block in response to the predicted excursion and based on an excursion limit and an impulse response model of the output transducer.

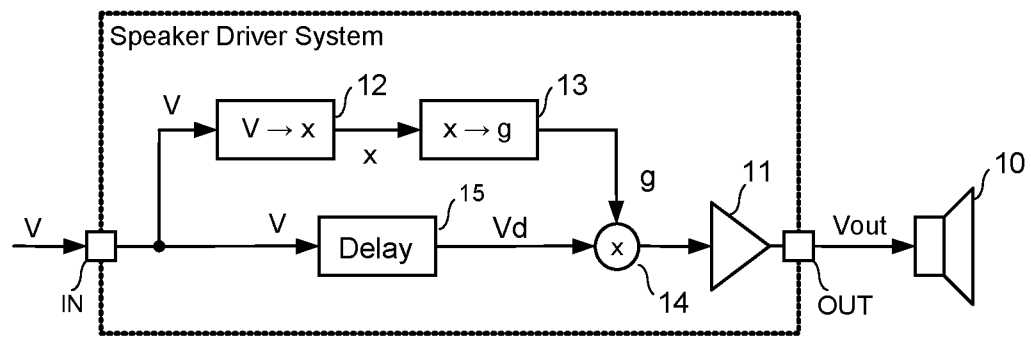


Figure 1

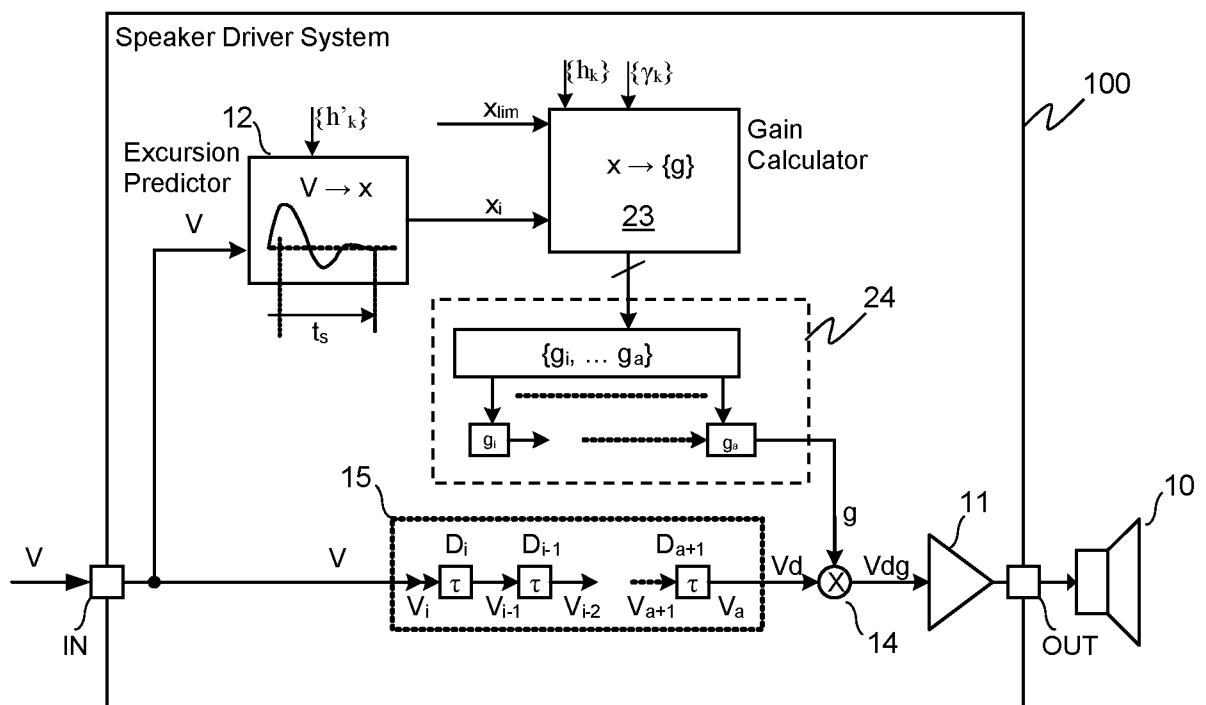


Figure 2

2/9

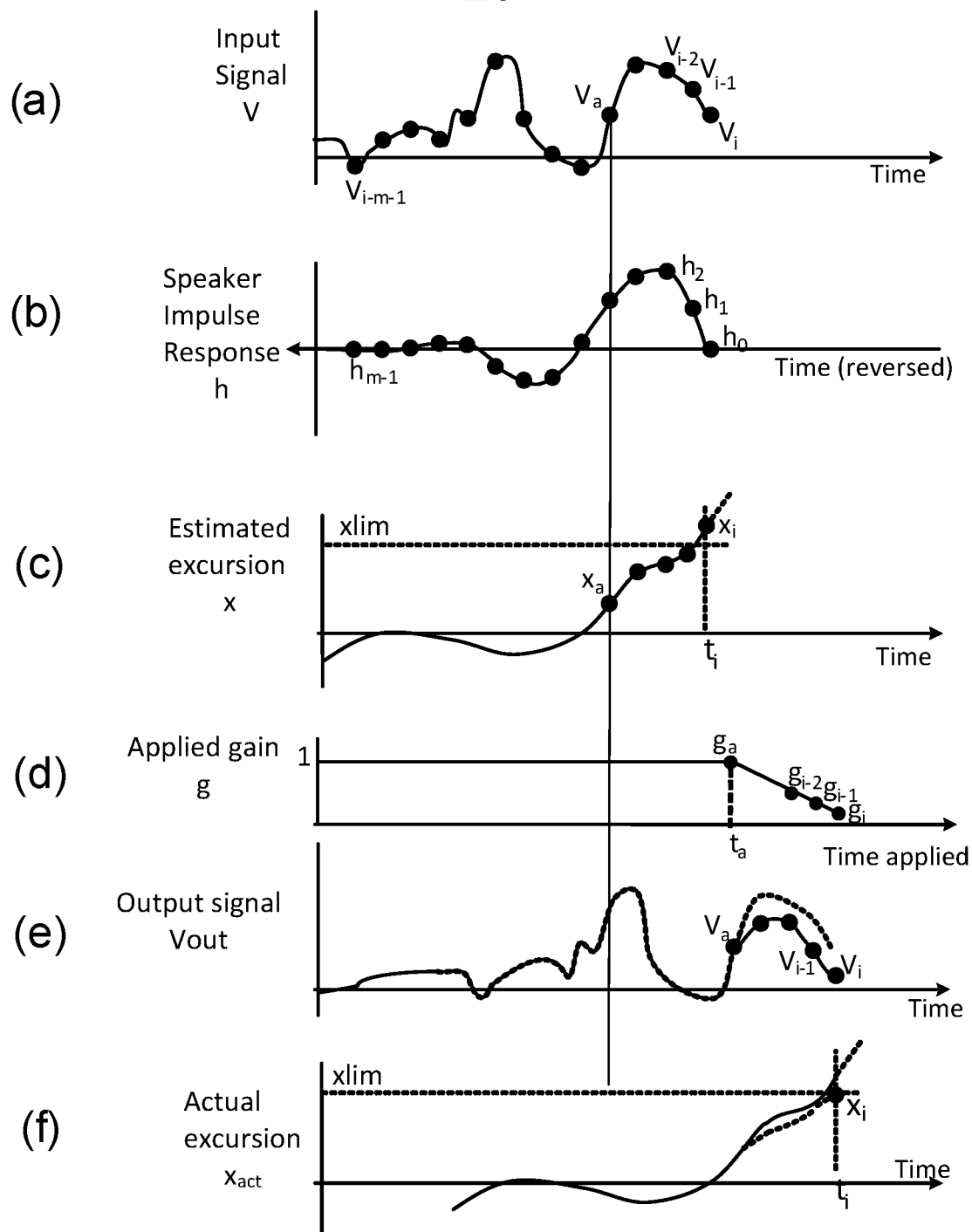


Figure 3

3/9

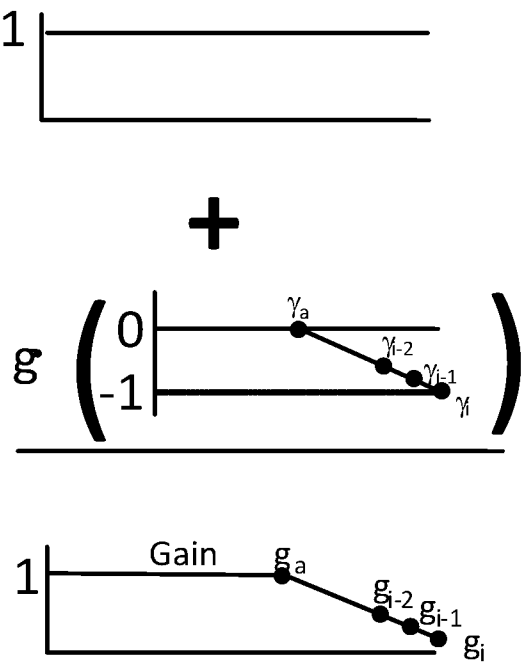


Figure 4

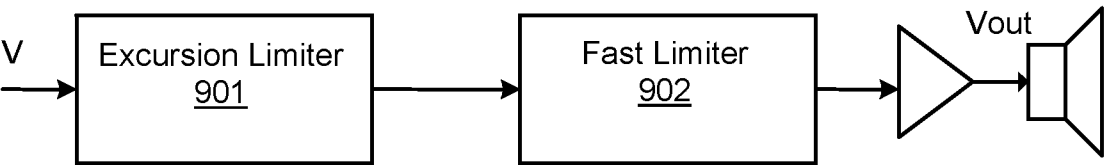


Figure 9

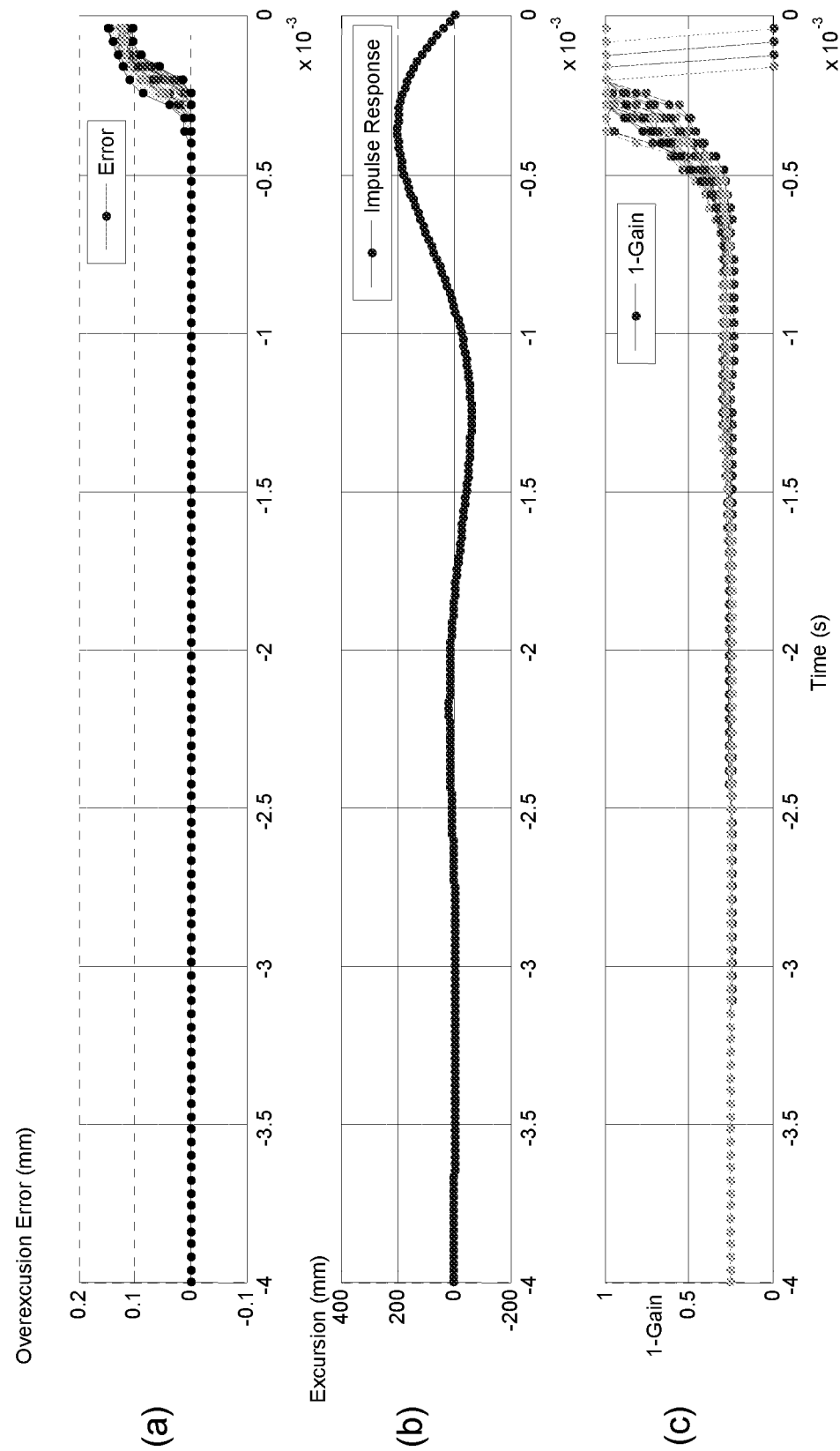


Figure 5

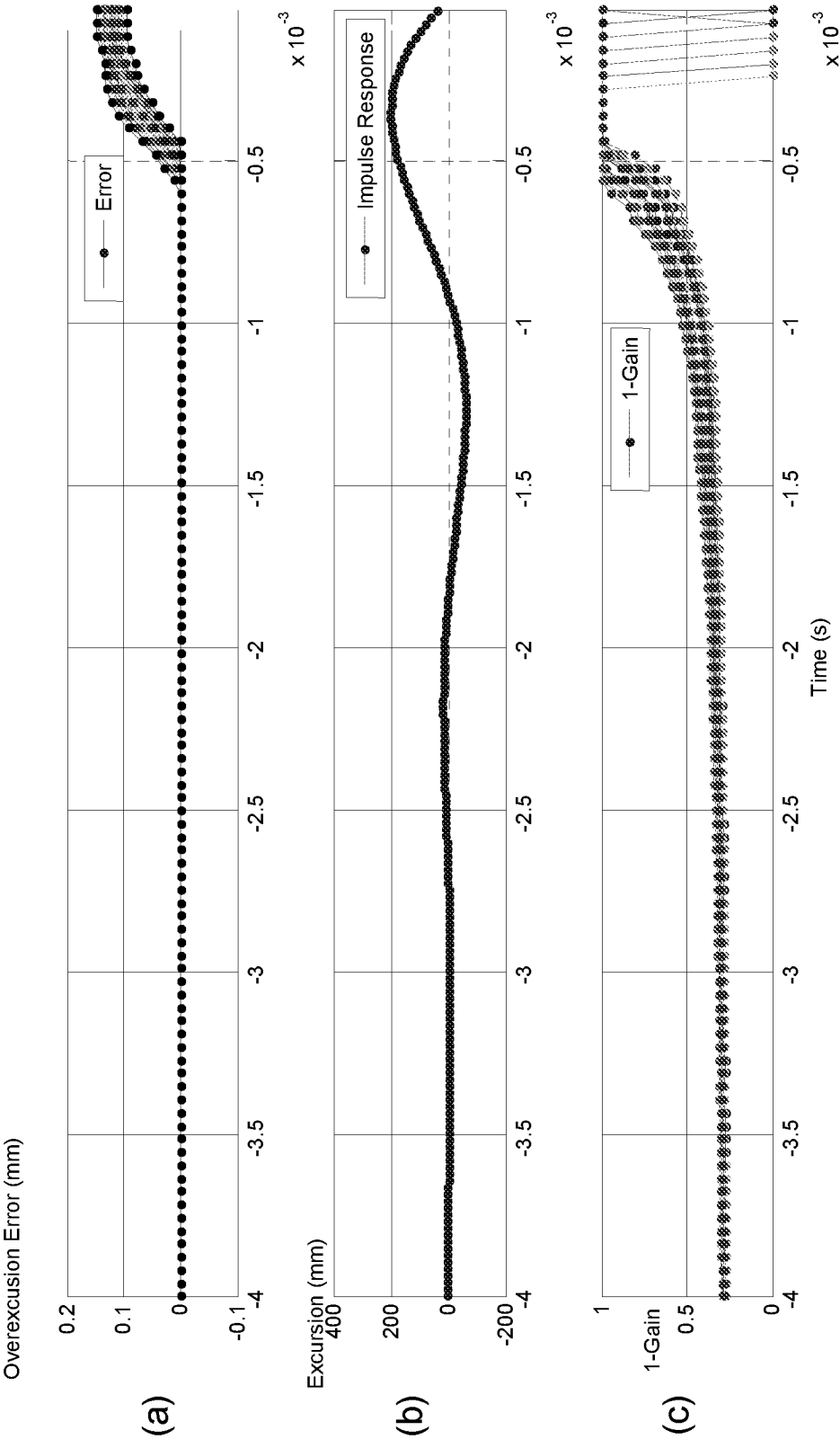


Figure 6

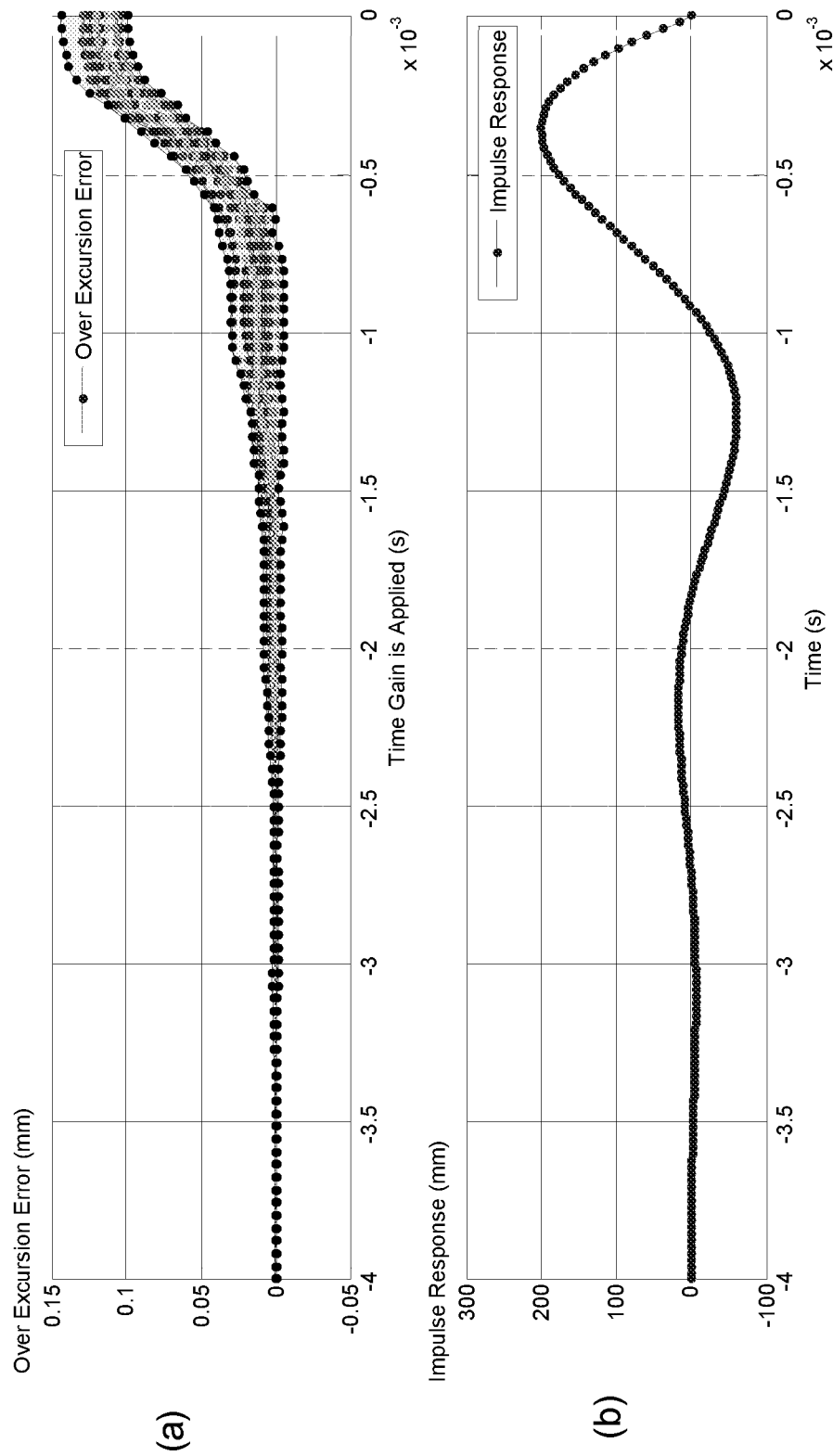


Figure 7

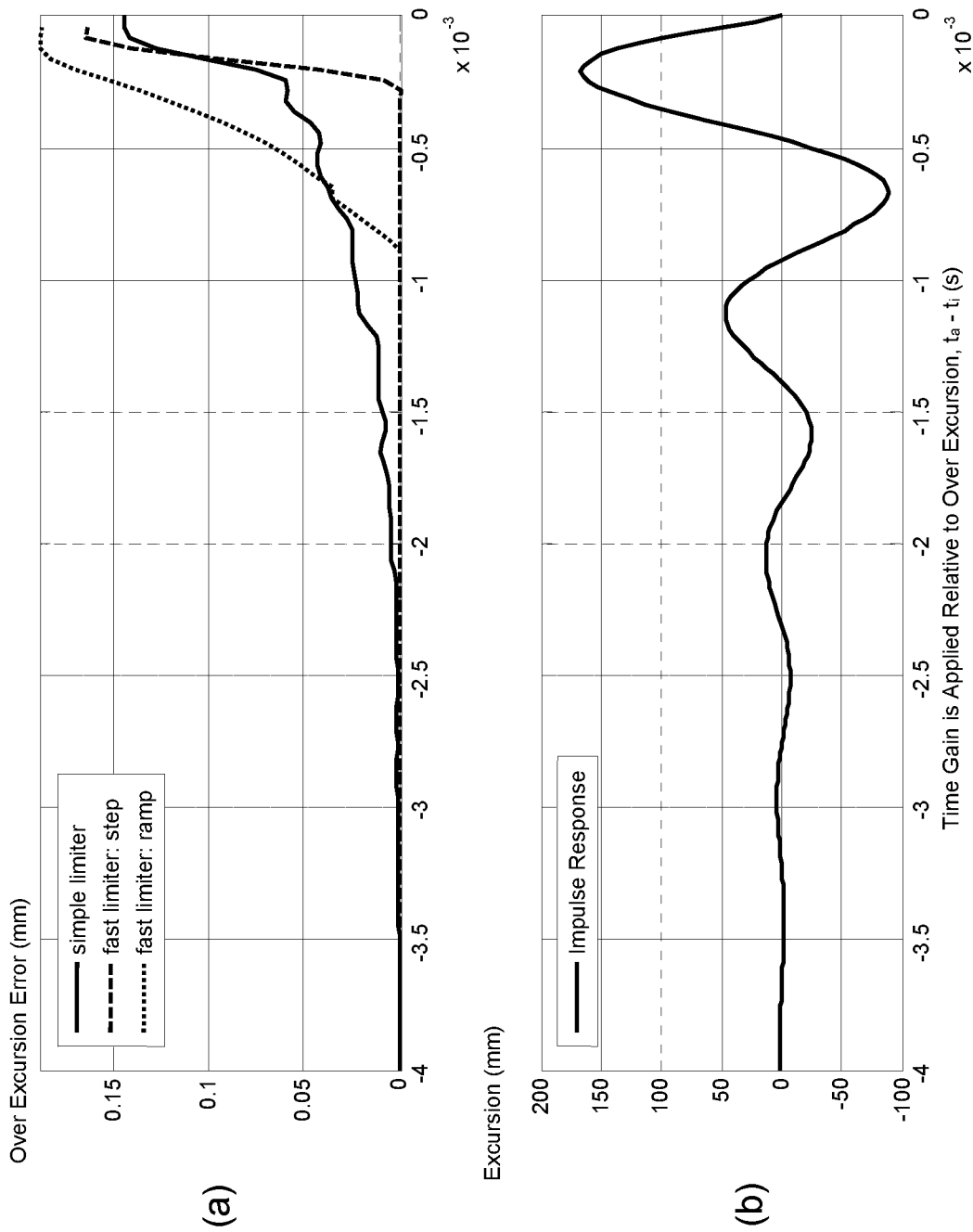


Figure 8

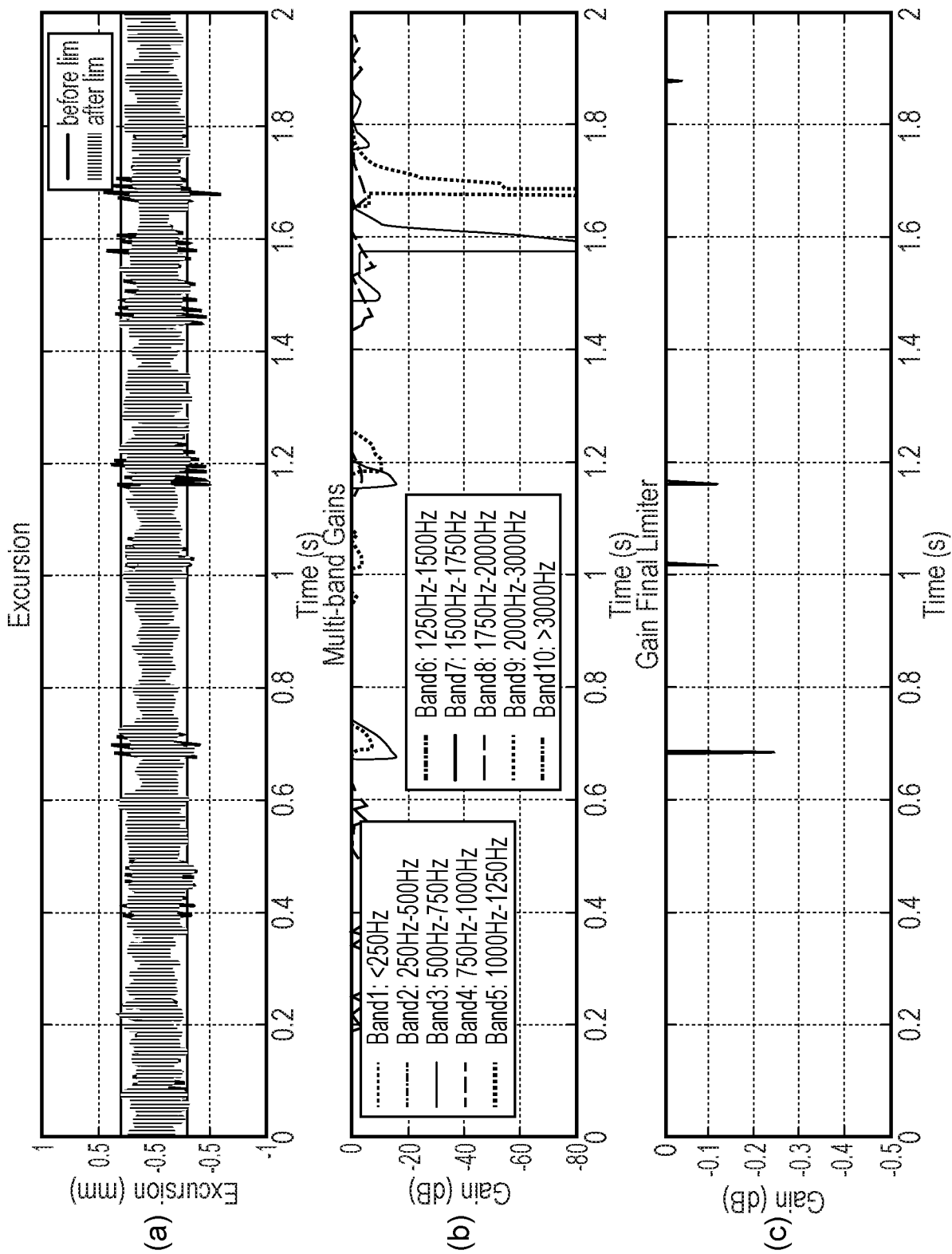


Figure 10

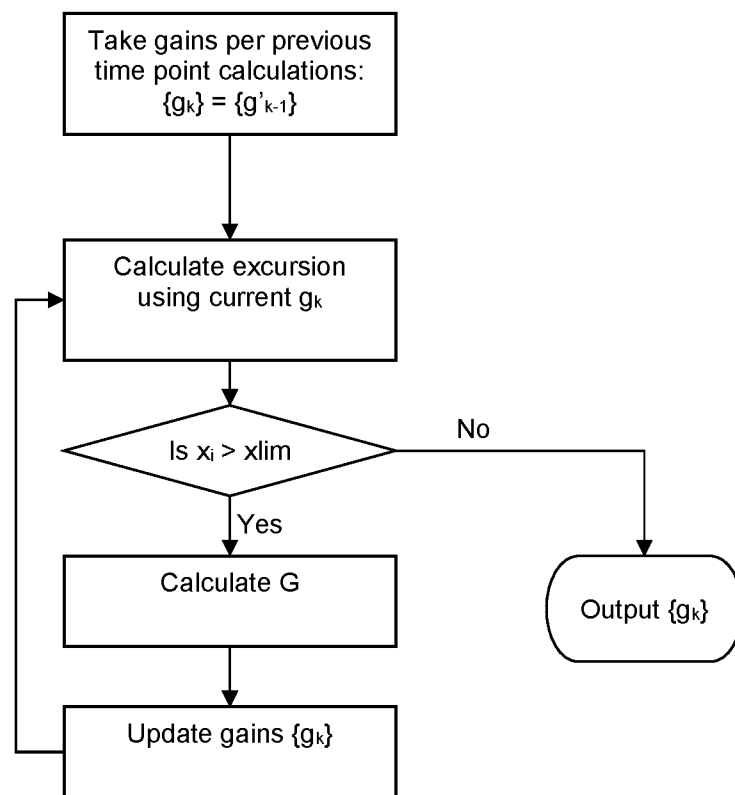


Figure 11

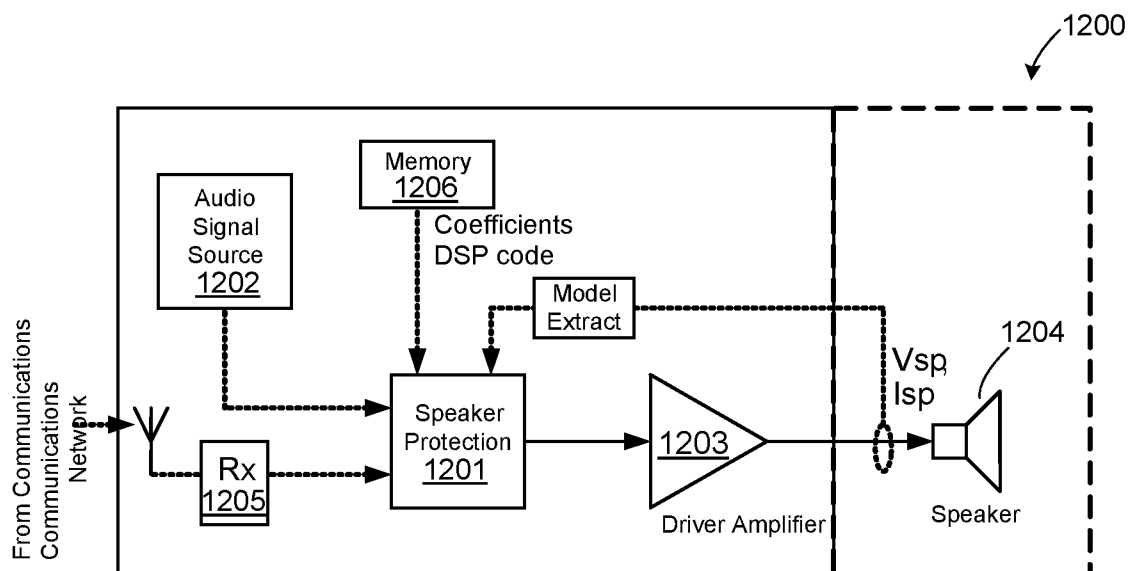


Figure 12

INTERNATIONAL SEARCH REPORT

International application No

PCT/GB2016/051850

A. CLASSIFICATION OF SUBJECT MATTER

INV. H04R3/00
ADD.

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

H04R H03G

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

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

EP0-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 2009/268918 A1 (SOLGAARD MADSEN [DK] ET AL) 29 October 2009 (2009-10-29) abstract paragraph [0044] - paragraph [0054]; claims 1,4; figure 2	1-33
X	US 9 060 217 B2 (THORMUNDSSON TRAUSTRUP [US] ET AL) 16 June 2015 (2015-06-16) abstract column 6, line 6 - column 8, line 46; figures 3,4	1
A	WO 2012/096897 A1 (QUALCOMM INC [US]; RYU SANG-UK [US]; SHIN JONGWON [US]; SILVERSTEIN RO) 19 July 2012 (2012-07-19) abstract; figure 1	1-33



Further documents are listed in the continuation of Box C.



See patent family annex.

* Special categories of cited documents :

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

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search

22 September 2016

Date of mailing of the international search report

30/09/2016

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Authorized officer

Gerken, Stephan

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No

PCT/GB2016/051850

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 2009268918	A1	29-10-2009	NONE
US 9060217	B2	16-06-2015	TW 201214954 A 01-04-2012
		US 2012106750 A1 03-05-2012	
		WO 2012009670 A2 19-01-2012	
WO 2012096897	A1	19-07-2012	CN 103299655 A 11-09-2013
		EP 2664161 A1 20-11-2013	
		JP 5763212 B2 12-08-2015	
		JP 2014506076 A 06-03-2014	
		US 2012179456 A1 12-07-2012	
		WO 2012096897 A1 19-07-2012	