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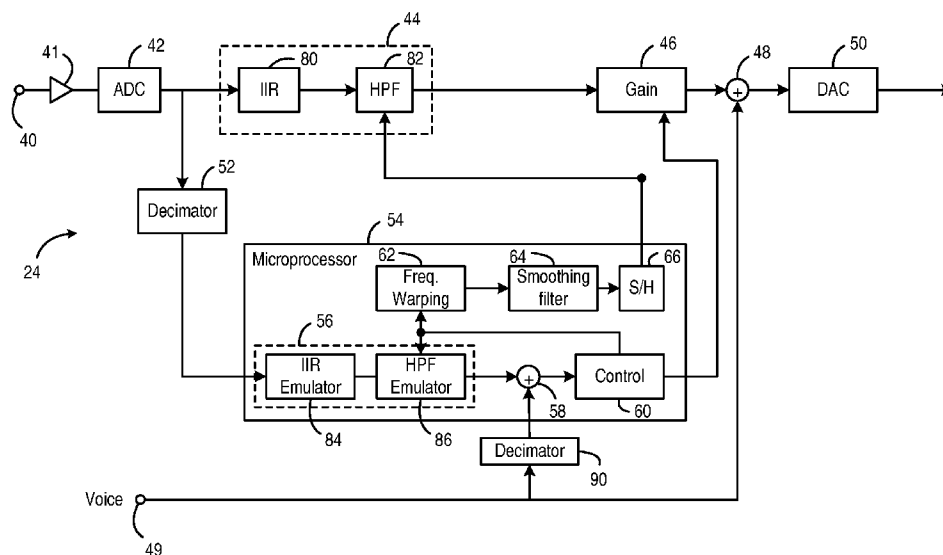


Figure 2

(57) Abstract: There is provided a noise cancellation system, comprising an input for a digital signal, the digital signal having a first sample rate; a digital filter, including a fixed filter and an adaptive filter; and control circuitry, for generating at least one control signal based on the input digital signal, wherein said at least one control signal is applied to the adaptive filter to control a filter characteristic thereof.

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NOISE CANCELLING SYSTEM WITH ADAPTIVE HIGH-PASS FILTER

This invention relates to a noise cancellation system, and in particular to a noise cancellation system having a filter that can easily be adapted based on an input signal
5 in order to improve the noise cancellation performance.

BACKGROUND

Noise cancellation systems are known, in which an electronic noise signal representing
10 ambient noise is applied to a signal processing circuit, and the resulting processed noise signal is then applied to a speaker, in order to generate a sound signal. In order to achieve noise cancellation, the generated sound should approximate as closely as possible the inverse of the ambient noise, in terms of its amplitude and its phase.

15 In particular, feedforward noise cancellation systems are known, for use with headphones or earphones, in which one or more microphones mounted on the headphones or earphones detect an ambient noise signal in the region of the wearer's ear. In order to achieve noise cancellation, the generated sound then needs to approximate as closely as possible the inverse of the ambient noise, after that ambient
20 noise has itself been modified by the headphones or earphones. One example of modification by the headphones or earphones is caused by the different acoustic path the noise must take to reach the wearer's ear, travelling around the edge of the headphones or earphones.

25 The microphone used to detect the ambient noise signal and the loudspeaker used to generate the sound signal from the processed noise signal will in practice also modify the signals, for example being more sensitive at some frequencies than at others. One example of this is when the speaker is closely coupled to the ear of a user, causing the frequency response of the loudspeaker to change due to cavity effects.

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Thus, the signal processing circuit should ideally be able to compensate for all of these effects. In order to be able to achieve this compensation, a relatively complex filter, for example a digital filter such as an infinite response (IIR) filter may be useful. However, it would be disadvantageous to have to perform full adaptation on a complex filter, such
35 as an IIR filter, in use of the device.

SUMMARY OF INVENTION

According to a first aspect of the present invention, there is provided a noise
5 cancellation system, comprising: an input for a digital signal, the digital signal having a
first sample rate; a digital filter, including a fixed filter and an adaptive high-pass filter;
and control circuitry, for generating at least one control signal based on the input digital
signal, wherein said at least one control signal is applied to the adaptive filter to control
a cut-off frequency thereof.

10 This has the advantage that it is possible to adapt the filter characteristics in use, in
dependence on the characteristic of the noise signal, but allows a relatively complex
filter to be used, while nevertheless allowing relatively simple control for adaptation
purposes.

15 According to a second aspect of the present invention, there is provided a method of
controlling a filter for a noise cancellation system. The method comprises the steps of
receiving a digital signal, the digital signal having a first sample rate; filtering the digital
signal with a fixed filter; and filtering the digital signal with an adaptive high-pass filter.
20 The adaptive high-pass filter is adapted based on the input digital signal to control a
cut-off frequency thereof.

BRIEF DESCRIPTION OF THE DRAWINGS

25 For a better understanding of the present invention, and to show more clearly how it
may be carried into effect, reference will now be made, by way of example, to the
following drawings, in which:

30 Figure 1 illustrates a noise cancellation system in accordance with an aspect of the
invention;

Figure 2 illustrates a signal processing circuit in accordance with an aspect of the
invention in the noise cancellation system of Figure 1; and

Figure 3 illustrates a signal processing circuit appropriate for use in a feedback noise cancellation system in accordance with the present invention.

DETAILED DESCRIPTION

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Figure 1 illustrates in general terms the form and use of a noise cancellation system in accordance with the present invention.

Specifically, Figure 1 shows an earphone 10, being worn on the outer ear 12 of a user 14. Thus, Figure 1 shows a supra-aural earphone that is worn on the ear, although it will be appreciated that exactly the same principle applies to circumaural headphones worn around the ear and to earphones worn in the ear such as so-called ear-bud phones. The invention is equally applicable to other devices intended to be worn or held close to the user's ear, such as mobile phones and other communication devices.

15

Ambient noise is detected by microphones 20, 22, of which two are shown in Figure 1, although any number more or less than two may be provided. Ambient noise signals generated by the microphones 20, 22 are combined, and applied to signal processing circuitry 24, which will be described in more detail below. In one embodiment, where the microphones 20, 22 are analogue microphones, the ambient noise signals may be combined by adding them together. Where the microphones 20, 22 are digital microphones, i.e. where they generate a digital signal representative of the ambient noise, the ambient noise signals may be combined alternatively, as will be familiar to those skilled in the art. Further, the microphones could have different gains applied to them before they are combined, for example in order to compensate for sensitivity differences due to manufacturing tolerances.

20

This illustrated embodiment of the invention also contains a source 26 of a wanted signal. For example, where the noise cancellation system is in use in an earphone, such as the earphone 10, that is intended to be able to reproduce music, the source 26 may be an inlet connection for a wanted signal from an external source such as a sound reproducing device. In other applications, for example where the noise cancellation system is in use in a mobile phone or other communication device, the source 26 may include wireless receiver circuitry for receiving and decoding radio frequency signals. In other embodiments, there may be no source, and the noise

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cancellation system may simply be intended to cancel the ambient noise for the user's comfort.

5 The wanted signal, if any, from the source 26 is applied through the signal processing circuitry 24 to a loudspeaker 28, which generates a sound signal in the vicinity of the user's ear 12. In addition, the signal processing circuitry 24 generates a noise cancellation signal that is also applied to the loudspeaker 28.

10 One aim of the signal processing circuitry 24 is to generate a noise cancellation signal, which, when applied to the loudspeaker 28, causes it to generate a sound signal in the ear 12 of the user that is the inverse of the ambient noise signal reaching the ear 12.

15 In order to achieve this, the signal processing circuitry 24 needs to generate from the ambient noise signals generated by the microphones 20, 22 a noise cancellation signal that takes into account the properties of the microphones 20, 22 and of the loudspeaker 28, and also takes into account the modification of the ambient noise that occurs due to the presence of the earphone 10.

20 Figure 2 shows in more detail the form of the signal processing circuitry 24. An input 40 is connected to receive an input signal, for example directly from the microphones 20, 22. This input signal is amplified in an amplifier 41 and the amplified signal is applied to an analog-digital converter 42, where it is converted to a digital signal. The digital signal is applied to an adaptive digital filter 44, and the filtered signal is applied to an adaptable gain device 46. Those skilled in the art will appreciate that in the case
25 where the microphones 20, 22 are digital microphones, wherein an analog-digital converter is incorporated into the microphone capsule and the input 40 receives a digital input signal, the analog-digital converter 42 is not required.

30 The resulting signal is applied to an adder 48, where it is summed with the wanted voice signal received from a second input 49, to which the source 26 may be connected.

35 Thus, the filtering and level adjustment applied by the filter 44 and the gain device 46 are intended to generate a noise cancellation signal that allows the detected ambient noise to be cancelled.

The output of the adder 48 is applied to a digital-analog converter 50, so that it can be passed to the loudspeaker 28.

As mentioned above, the noise cancellation signal is produced from the input signal by the adaptive digital filter 44 and the adaptable gain device 46. These are controlled by one or more control signals, which are generated by applying the digital signal output from the analog-digital converter 42 to a decimator 52 which reduces the digital sample rate, and then to a microprocessor 54. In a preferred embodiment, the sample rate of the input digital signal is 2.4 MHz, and the decimator reduces this sample rate to 8 kHz.

10

The microprocessor 54 contains a block 56 that emulates the filter 44 and gain device 46, and produces an emulated filter output which is applied to an adder 58, where it is summed with the wanted signal from the second input 49, via a decimator 90. The sample rate reduction performed by the decimator 52 allows the emulation to be performed with lower power consumption than performing the emulation at the original 2.4 MHz sample rate.

15

The resulting signal is applied to a control block 60, which generates control signals for adjusting the properties of the filter 44 and the gain device 46. The control signal for the filter 44 is applied through a frequency warping block 62, a smoothing filter 64 and sample-and-hold circuitry 66 to the filter 44. The same control signal is also applied to the block 56, so that the emulation of the filter 44 matches the adaptation of the filter 44 itself.

20

The purpose of the frequency warping block 62 is to adapt the control signal output from the control block 60 for the high-frequency adaptive filter 82. That is, the high-frequency filter 82 will generally be operating at a frequency that is much higher than that of the low-frequency filter emulator 86, and therefore the control signal will generally need to be adapted in order to be applicable to both filters.

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In an alternative embodiment, the sample-and-hold circuitry 66 may be replaced by an interpolation filter.

The control block 60 further generates a control signal for the adaptive gain device 46. In the illustrated embodiment, the gain control signal is output directly to the gain device 46.

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As discussed above, the digital signal representing the detected ambient noise is applied to an adaptive digital filter 44, in order to generate a noise cancellation signal. In order to be able to use the signal processing circuitry 24 in a range of different applications, it is necessary for the adaptive digital filter 44 to be relatively complex, so that it can compensate for different microphone and speaker combinations, and for different types of earphone having different effects on the ambient noise.

However, it would be disadvantageous to have to perform full adaptation on a complex filter, such as an IIR filter, in use of the device. Thus, in this preferred embodiment of the invention, the filter 44 includes an IIR filter 80 having a filter characteristic that is effectively fixed while the device is in operation. More specifically, the IIR filter may have several possible sets of filter coefficients, the filter coefficients together defining the filter characteristic, with one of these sets of filter coefficients being applied based on the microphone 20, 22, speaker 28, and earphone 10 with which the signal processing circuitry 24 is being used.

The setting of the IIR filter coefficients may take place when the device is manufactured, or when the device is first inserted in a particular earphone 10, or as a result of a calibration process that occurs on initial power-up of the device or at periodic intervals (such as once per day, for example). Thereafter, the filter coefficients are not changed, and the filter characteristic is fixed, rather than being adapted on the basis of the signal being applied thereto.

However, it has been found that this may have the disadvantage that the device may not perform optimally under all conditions. For example, in situations where there is a relatively high level of low frequency noise, the resulting noise cancellation signal would be at a level that is higher than could be handled by a typical speaker 28.

Thus, the filter 44 also includes an adaptive component, in this illustrated example an adaptive high-pass filter 82. The properties of the high-pass filter, such as its cut-off frequency, can then be adjusted on the basis of the control signal generated by the microprocessor 54. Moreover, the adaptation of the filter 44 can then take place on the basis of a much simpler control signal.

The use of a filter that contains a fixed part and an adaptive part therefore allows for the use of a relatively complex filter, but allows for the adaptation of that filter by means of a relatively simple control signal.

5 As described so far, the adaptation of the filter 44 takes place on the basis of a control signal that is derived from the input to the filter. However, it is also possible that the adaptation of the filter 44 could take place on the basis of a control signal that is derived from the filter output. Moreover, the division of the filter into a fixed part and an adaptive part allows for the possibility that the adaptation of the filter 44 could take
10 place on the basis of a control signal that is derived from the output of the first of these filter stages. In particular, where, as illustrated, the signal is applied first to the fixed filter stage 80 and then to the adaptive filter stage 82, the adaptation of the adaptive filter stage 82 could take place on the basis of a control signal that is derived from the output of the fixed filter stage 80.

15 As mentioned above, the control signal is generated by a microprocessor 54 which contains an emulation of the filter 44. Therefore, where the filter 44 contains a fixed stage 80 and an adaptive stage 82, the emulation 56 should preferably also contain a fixed stage 84 and an adaptive stage 86, so that it can be adapted in the same way.

20 Further, the microprocessor 54 may comprise an adaptive gain emulator (not shown in Figure 2), located in between the filter emulator 56 and the adder 58. In this instance, the control block 60 will also output the gain control signal to the adaptive gain emulator.

25 In the illustrated embodiment described so far, the fixed filter 80 is an IIR filter, but it will be appreciated that the fixed filter could be any digital filter, for example such as a finite impulse response (FIR) filter. However, in preferred embodiments, the fixed filter stage 80 is such that it has a filter characteristic that is generally not flat over the frequency
30 range of interest (which may for example be 50 Hz to 5 kHz), but rather suppresses signals in certain parts of that frequency range more than signals in certain other parts of that frequency range.

Also, in the illustrated embodiment the adaptive filter stage 82 is a first-order high pass
35 filter, having a frequency characteristic that is generally flat above a corner frequency, and reducing below that corner frequency, with the corner frequency being determined

by the control signal from the microprocessor 54. However, the adaptive filter stage can in principle take any convenient form, although the advantage of the present invention is most noticeable when the adaptive filter stage 82 is relatively simple, with only a small number of filter parameters being adaptive. In an embodiment where the
5 adaptive filter stage 82 is more complex, the microprocessor 54 may generate a plurality of control signals to change the coefficients of the adaptive filter stage 82.

Various modifications may be made to the embodiments described above without departing from the scope of the claims appended hereto. For example, the voice signal
10 input to the signal processor 24 may be digital, as described above, or analog, in which case an analog-digital converter may be necessary to convert the signal to digital.

Further, the fixed filter 180 and adaptive filter 182 may be arranged so that the adaptive filter 182 is located prior to the fixed filter 180.

15 It will be apparent to those skilled in the art that the present invention is equally applicable to so-called feedback noise cancellation systems.

The feedback method is based upon the use, inside the cavity that is formed between
20 the ear and the inside of an earphone shell, or between the ear and a mobile phone, of a microphone placed directly in front of the loudspeaker. Signals derived from the microphone are coupled back to the loudspeaker via a negative feedback loop (an inverting amplifier), such that it forms a servo system in which the loudspeaker is constantly attempting to create a null sound pressure level at the microphone.

25 Figure 3 shows an example of signal processing circuitry according to the present invention when implemented in a feedback system.

The feedback system comprises a microphone 120 positioned substantially in front of a
30 loudspeaker 128. The microphone 120 detects the output of the loudspeaker 128, with the detected signal being fed back via an amplifier 141 and an analog-to-digital converter 142. A wanted audio signal is fed to the processing circuitry via an input 140. The fed back signal is subtracted from the wanted audio signal in a subtracting element 188, in order that the output of the subtracting element 188 substantially represents the
35 ambient noise, i.e. the wanted audio signal has been substantially cancelled.

Thereafter, the processing circuitry is substantially similar to the processing circuitry 24 in the feed forward system described with respect to Figure 2. The output of the subtracting element 188 is fed to an adaptive digital filter 144, and the filtered signal is applied to an adaptable gain device 146.

5

The resulting signal is applied to an adder 148, where it is summed with the wanted audio signal received from the input 140.

Thus, the filtering and level adjustment applied by the filter 144 and the gain device 146 are intended to generate a noise cancellation signal that allows the detected ambient noise to be cancelled.

The output of the adder 148 is applied to a digital-analog converter 150, so that it can be passed to the loudspeaker 128.

15

As mentioned above, the noise cancellation signal is produced from the input signal by the adaptive digital filter 144 and the adaptable gain device 146. These are controlled by a control signal, which is generated by applying the digital signal output from the analog-digital converter 142 to a decimator 152 which reduces the digital sample rate, and then to a microprocessor 154.

20

The microprocessor 154 contains a block 156 that emulates the filter 144 and gain device 146, and produces an emulated filter output which is applied to an adder 158, where it is summed with the wanted audio signal from the input 140 via a decimator 190.

25

The resulting signal is applied to a control block 160, which generates control signals for adjusting the properties of the filter 144 and the gain device 146. The control signal for the filter 144 is applied through a frequency warping block 162, a smoothing filter 164 and sample-and-hold circuitry 166 to the filter 144. The same control signal is also applied to the block 156, so that the emulation of the filter 144 matches the adaptation of the filter 144 itself.

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In an alternative embodiment, the sample-and-hold circuitry 166 is replaced by an interpolation filter.

35

The control block 160 further generates a control signal for the adaptive gain device 146. In the illustrated embodiment, the gain control signal is output directly to the gain device 146.

- 5 Further, the microprocessor 154 may comprise an adaptive gain emulator (not shown in Figure 3), located in between the filter emulator 156 and the adder 158. In this instance, the control block 160 will also output the gain control signal to the adaptive gain emulator.
- 10 Similarly to the feedforward case, the fixed filter 180 may be an IIR filter, and the adaptive filter 182 may be a high pass filter.

It will be clear to those skilled in the art that the implementation may take one of several hardware or software forms, and the intention of the invention is to cover all
15 these different forms.

Noise cancellation systems according to the present invention may be employed in many devices, as would be appreciated by those skilled in the art. For example, they may be employed in mobile phones, headphones, earphones, headsets, etc.

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The skilled person will recognise that the above-described apparatus and methods may be embodied as processor control code, for example on a 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
25 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-
30 programmable logic gate arrays. Similarly the code may comprise code for a hardware description language such as Verilog TM 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
35 running on a field-(re-)programmable analogue array or similar device in order to configure analogue/digital hardware.

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

5 “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 processor or other unit may fulfil the functions of several units recited in the claims. Any reference signs in the claims shall not be construed so as to limit their scope.

CLAIMS

1. A noise cancellation system, comprising:
an input for a digital signal, the digital signal having a first sample rate;
5 a digital filter, including a fixed filter and an adaptive high-pass filter; and
control circuitry, for generating at least one control signal based on the input
digital signal, wherein said at least one control signal is applied to the adaptive filter to
control a cut-off frequency thereof.
- 10 2. A noise cancellation system as claimed in claim 1, further comprising:
an adaptive gain element;
wherein the control circuitry is adapted to generate a second control signal for
application to the adaptive gain element to control a gain thereof.
- 15 3. A noise cancellation system as claimed in any preceding claim, wherein the fixed
filter is an IIR filter.
4. A noise cancellation system as claimed in any preceding claim, further
comprising:
20 a decimator, for receiving the digital signal and generating a decimated signal at
a second sample rate lower than the first sample rate, wherein the decimated signal is
applied to the control circuitry.
5. A noise cancellation system as claimed in any preceding claim, further
25 comprising:
a second input for receiving a wanted signal, the second input being connected
to the control circuitry, and the control circuitry being adapted to generate a control
signal based also on the input wanted signal.
- 30 6. A noise cancellation system as claimed in one of claims 1 to 5, wherein the
control circuitry is adapted to generate said at least one control signal based on the
unfiltered input digital signal.
7. A noise cancellation system as claimed in one of claims 1 to 5, wherein the
35 control circuitry is adapted to generate said at least one control signal based on an
output of the fixed filter.

8. A noise cancellation system as claimed in one of claims 1 to 5, wherein the control circuitry is adapted to generate said at least one control signal based on an output of the adaptive high-pass filter.
- 5
9. A noise cancellation system as claimed in any one of claims 1 to 8, wherein said noise cancellation system is a feedforward system.
10. A noise cancellation system as claimed in any one of claims 1 to 8, wherein said
- 10 noise cancellation system is a feedback system.
11. An integrated circuit, comprising:
a noise cancellation system as claimed in any one of claims 1 to 10.
- 15
12. A mobile phone, comprising:
an integrated circuit as claimed in claim 11.
13. A pair of headphones, comprising:
an integrated circuit as claimed in claim 11.
- 20
14. A pair of earphones, comprising:
an integrated circuit as claimed in claim 11.
15. A headset, comprising:
- 25 an integrated circuit as claimed in claim 11.
16. A method of controlling a filter for a noise cancellation system, comprising:
receiving a digital signal, the digital signal having a first sample rate;
filtering the digital signal with a fixed filter; and
- 30 filtering the digital signal with an adaptive high-pass filter;
wherein the adaptive high-pass filter is adapted based on the input digital signal to control a cut-off frequency thereof.
17. A method as claimed in claim 16, further comprising:
- 35 generating a decimated signal from the input digital signal, said decimated signal having a second sample rate lower than the first sample rate;

wherein the adaptive filter is adapted based on the decimated signal.

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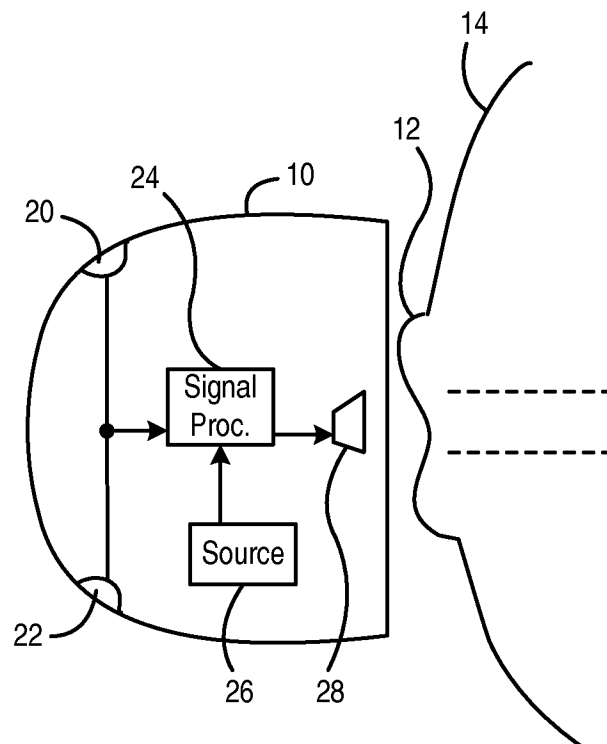


Figure 1

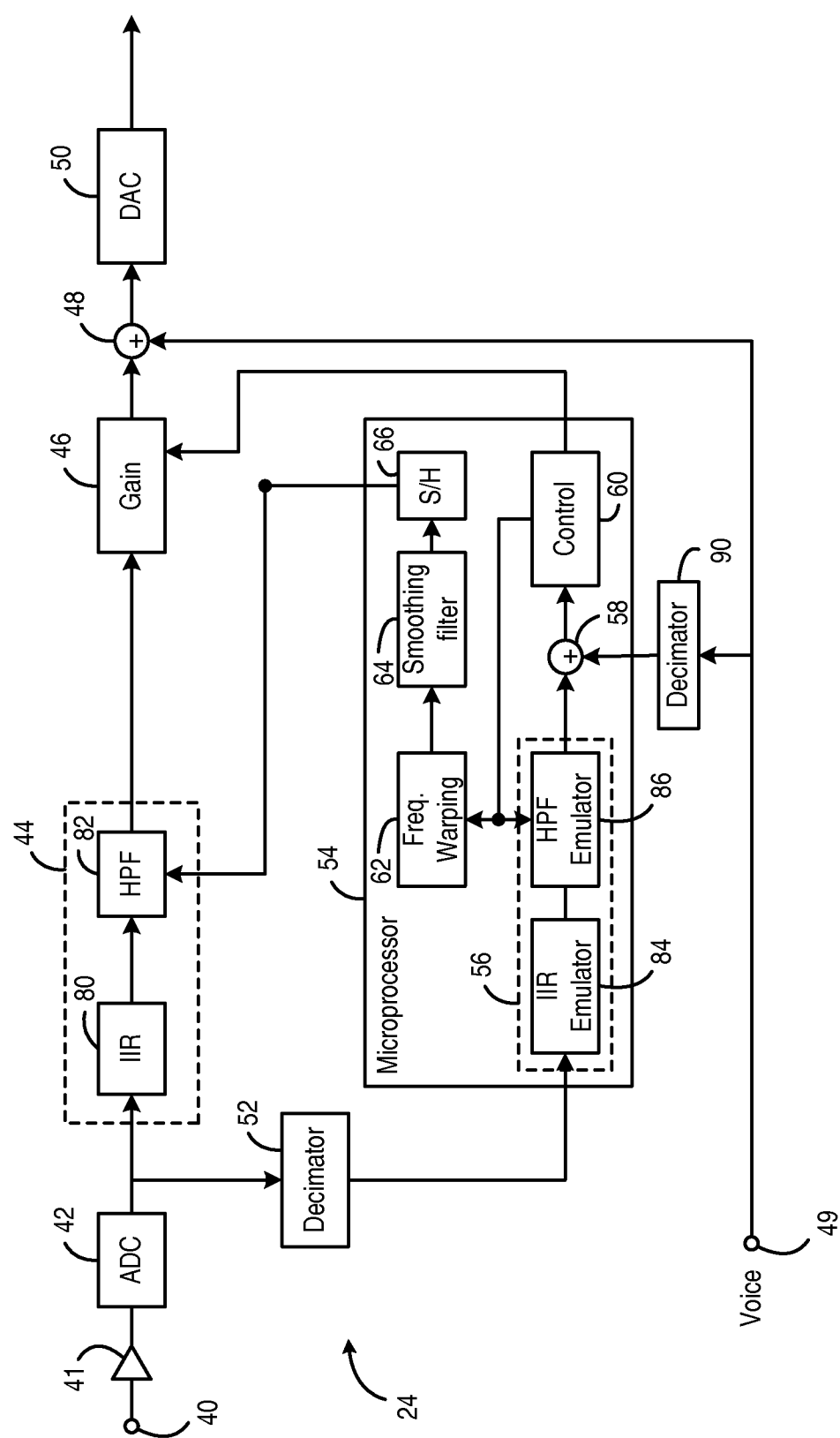


Figure 2

INTERNATIONAL SEARCH REPORT

International application No
PCT/GB2008/051189

A. CLASSIFICATION OF SUBJECT MATTER
INV. G10K11/178

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)

G10K H04R

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

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

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	JP 2005 257720 A (MATSUSHITA ELECTRIC IND CO LTD) 22 September 2005 (2005-09-22) abstract; figures 2,17	1-17
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A	EP 1 397 024 A (SONY ERICSSON MOBILE COMM AB [SE]) 10 March 2004 (2004-03-10) paragraphs [0029] - [0032]; figures 1,2	1,16

☐ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

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Date of the actual completion of the international search

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Name and mailing address of the ISA/

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No

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