Abstract Title: Noise cancellation system with adaptive filter and two different sample rates

The system and method comprising: an adaptive filter 44 for receiving a digital noise signal 40 at a first sample rate and for generating a noise cancellation signal; and control circuitry 54, for generating a control signal at a second sample rate for application to the adaptive filter so as to adjust a filter characteristic. The second sample rate is lower than the first sample rate but is higher than a Nyquist sampling rate required to sample signals up to a desired cut-off frequency within the audio frequency range. The system may employ analog-digital converters 42 if the signal sources are not digital, and may use a decimator 52 to reduce signal sample rates. The first sample rate may vary from 100 kHz to over 1 MHz and the second sample rate from 50 kHz to less than 2 kHz. The system may find application in mobile phones, headphones, earphones etc..
SLOW RATE ADAPTATION

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 in order to improve the noise cancellation performance.

BACKGROUND

Noise cancellation systems are known, in which an electronic noise signal representing 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.

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 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.

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.

It is therefore advantageous to provide an adaptive filter, in order to be able to adapt the signal processing to the present form of the ambient noise signal.

SUMMARY OF INVENTION
In accordance with a first aspect of the present invention, there is provided a noise cancellation system, comprising: an adaptive filter, for receiving a digital noise signal at a first sample rate and generating a noise cancellation signal; and control circuitry, for generating a control signal having a second sample rate for application to the adaptive filter to adjust a filter characteristic thereof, wherein the second sample rate is lower than the first sample rate but is higher than a Nyquist sampling rate required to sample signals up to a desired cut-off frequency within the audio frequency range.

This has the advantage that the filter can operate at a very high sample rate, while being adapted at a slower rate, which is nevertheless high enough to allow the adaptation to take full account of any changes which may produce audible effects.

**BRIEF DESCRIPTION OF THE DRAWINGS**

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:

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**

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.

5 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.

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

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.

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.
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 (for example, as 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).

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 adaptable 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 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.

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.

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 adaptable digital filter 44 and the adaptable gain device 46. These are controlled by a control signal, which is 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.
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.

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.

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.

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.

In this illustrated embodiment of the invention, the filter 44 comprises a fixed IIR filter 80 and an adaptive high-pass filter 82, and the filter emulation 56 similarly comprises a fixed IIR filter 84 and an adaptive high-pass filter 86, which either mirror, or are sufficiently accurate approximations of, the filters which they emulate.

However, the invention may be applied to any filter arrangement, in which the filter comprises a filter stage or multiple filter stages, provided that at least one such stage is
adaptive. Moreover, the filter may be relatively complex, such as an IIR filter, or may be relatively simple, such as a low-order low-pass or high-pass filter.

Further, the possible filter adaptation may be relatively complex, with several different parameters being adaptive, or may be relatively simple, with just one parameter being adaptive. For example, in the illustrated embodiment, the adaptive high-pass filter 82 is a first-order filter controllable by a single control value, which has the effect of altering the filter corner frequency. However, in other cases the adaptation may take the form of altering several parameters of a higher order filter, or may in principle take the form of altering the full set of filter coefficients of an IIR filter.

It is well known that, in order to process digital signals, it is necessary to operate with signals that have a sample rate that is at least twice the frequency of the information content of the signals, and that signal components at frequencies higher than half the sampling rate will be lost. In a situation where signals at frequencies up to a cut-off frequency must be handled, there is thus defined the Nyquist sampling rate, which is twice this cut-off frequency.

A noise cancellation system is generally intended to cancel only audible effects. As the upper frequency of human hearing is typically 20 kHz, this would suggest that acceptable performance could be achieved by sampling the noise signal at a sampling rate in the region of 40 kHz. However, in order to achieve adequate performance, this would require sampling the noise signal with a relatively high degree of precision, and there would inevitably be delays in the processing of such signals.

In the illustrated embodiment of the invention, therefore, the analog-digital converter 42 generates a digital signal at a sample rate of 2.4 MHz, but with a bit resolution of only 3 bits. This allows for acceptably accurate signal processing, but with much lower signal processing delays. In other embodiments of the invention, the sample rate of the digital signal may be 44.1 kHz, or greater than 100 kHz, or greater than 300 kHz, or greater than 1 MHz.

As described above, the filter 44 is adaptive. That is, a control signal can be sent to the filter to change its properties, such as its frequency characteristic. In the illustrated embodiment of this invention, the control signal is sent not at the sampling rate of the
digital signal, but at a lower rate. This saves power and processing complexity in the control circuitry, in this case the microprocessor 54.

The control signal is sent at a rate that allows it to adapt the filter sufficiently quickly to handle changes that may possibly produce audible effects, namely at least equal to the Nyquist sampling rate defined by a desired cut-off frequency in the audio frequency range.

Although it would be desirable to be able to achieve noise cancellation across the whole of the audio frequency range, in practice it is usually only possible to achieve good noise cancellation performance over a part of the audio frequency range. In a typical case, it is considered preferable to optimize the system to achieve good noise cancellation performance over the lower part of the audio frequency range, for example from 80 Hz to 2.5 kHz. It is therefore sufficient to generate a control signal having a sample rate which is twice the frequency above which it is not expected to achieve outstanding noise cancellation performance.

In the illustrated embodiment of the invention, the control signal has a sampling rate of 8 kHz, but, in other embodiments of the invention, the control signal may have a sampling rate which is less then 2 kHz, or less than 10 kHz, or less than 20 kHz, or less than 50 kHz.

In the illustrated embodiment of the invention, the decimator 52 reduces the sample rate of the digital signal from 2.4 MHz to 8 kHz, and the microprocessor 54 produces a control signal at the same sampling rate as its input signal. However, the microprocessor 54 can in principle produces a control signal having a sampling rate that is higher, or lower, than its input signal received from the decimator 52.

The illustrated embodiment shows the noise signal being received from an analog source, such as a microphone, and being converted to digital form in an analog-digital converter 42 in the signal processing circuitry. However, it will be appreciated that the noise signal could be received in a digital form, from a digital microphone, for example.

Further, the illustrated embodiment shows the noise cancellation signal being generated in a digital form, and being converted to analog form in a digital-analog
converter 50 in the signal processing circuitry. However, it will be appreciated that
the noise cancellation signal could be output in a digital form, for example for application to
a digital speaker, or the like.

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.

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
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.

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
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
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.

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.

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.

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.

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.

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

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.

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.

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
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.

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
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 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
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
"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 adaptive filter, for receiving a digital noise signal at a first sample rate and
   generating a noise cancellation signal; and
   control circuitry, for generating a control signal having a second sample rate for
   application to the adaptive filter to adjust a filter characteristic thereof,
   wherein the second sample rate is lower than the first sample rate but is higher
   than a Nyquist sampling rate required to sample signals up to a desired cut-off
   frequency within the audio frequency range.

2. A noise cancellation system as claimed in claim 1, further comprising:
   an analog-digital converter, for receiving an input signal representative of
   ambient noise, and generating the digital noise signal at the first sample rate;

3. A noise cancellation system as claimed in claim 1 or 2, further comprising:
   a decimator, for receiving the digital noise signal and generating a decimated
   noise signal, and applying the decimated noise signal to the control circuitry.

4. A noise cancellation system as claimed in claim 3, wherein the decimator
   generates the decimated noise signal at the second sample rate.

5. A noise cancellation system as claimed in any preceding claim, wherein the
   control circuitry is connected to receive a wanted signal, and is adapted to generate the
   control signal based also on the received wanted signal.

6. A noise cancellation system as claimed in any preceding claim, further
   comprising:
   a digital-analog converter, for receiving the noise cancellation signal and
   converting it to an analog noise cancellation signal.

7. A noise cancellation system as claimed in any preceding claim, wherein the first
   sample rate is greater than 100 kHz.
8. A noise cancellation system as claimed in claim 7, wherein the first sample rate is greater than 300 kHz.

9. A noise cancellation system as claimed in claim 8, wherein the first sample rate is greater than 1 MHz.

10. A noise cancellation system as claimed in any preceding claim, wherein the second sample rate is less than 50 kHz.

11. A noise cancellation system as claimed in claim 10, wherein the second sample rate is less than 20 kHz.

12. A noise cancellation system as claimed in claim 11, wherein the second sample rate is less than 10 kHz.

13. A noise cancellation system as claimed in claim 12, wherein the second sample rate is less than 2 kHz.

14. A noise cancellation system as claimed in any one of the preceding claims, wherein the noise cancellation system is a feedforward noise cancellation system.

15. A noise cancellation system as claimed in any one of claims 1 to 13, wherein the noise cancellation system is a feedback noise cancellation system.

16. An integrated circuit, comprising:
   a noise cancellation system as claimed in any one of claims 1 to 15.

17. A mobile phone, comprising:
   an integrated circuit as claimed in claim 16.

18. A pair of headphones, comprising:
   an integrated circuit as claimed in claim 16.

19. A pair of earphones, comprising:
   an integrated circuit as claimed in claim 16.
20. A headset, comprising:
    an integrated circuit as claimed in claim 16.

5  21. A method of controlling a filter for a noise cancellation system, comprising:
    receiving a digital noise signal, the digital noise signal having a first sample rate;
    filtering the digital noise signal in an adaptive filter to generate a noise cancellation signal; and
    generating a control signal having a second sample rate for application to the adaptive filter to adjust a filter characteristic thereof,
    wherein the second sample rate is lower than the first sample rate but is higher than a Nyquist sampling rate required to sample signals up to a desired cut-off frequency within the audio frequency range.

15  22. A method as claimed in claim 21, further comprising:
    receiving an input signal representative of ambient noise; and
    generating the digital noise signal at the first sample rate.

23. A method as claimed in claim 21 or 22, further comprising:
    generating a decimated noise signal; and
    generating the control signal from the decimated noise signal.

24. A method as claimed in claim 23, wherein the decimated noise signal is generated at the second sample rate.
Application No: GB0811001.7  Examiner: Mr Alan Phipps
Claims searched: 1-24  Date of search: 16 October 2008

Patents Act 1977: Search Report under Section 17

Documents considered to be relevant:

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<th>Category</th>
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Field of Search:

Search of GB, EP, WO & US patent documents classified in the following areas of the UKC⁺:

Worldwide search of patent documents classified in the following areas of the IPC
G10K; H04L

The following online and other databases have been used in the preparation of this search report
WPI, EPDOC, IEEE

International Classification:

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