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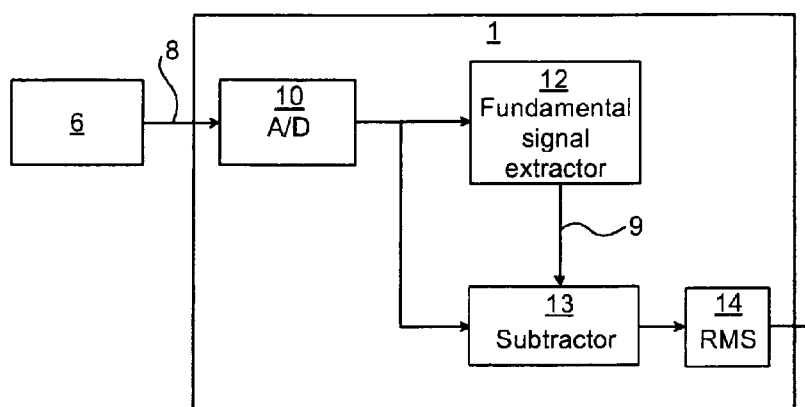


Fig. 1

(57) Abstract: It is presented a harmonics detector (1) for detecting harmonics of a signal source (8). The harmonics detector comprises: a fundamental extractor (12) comprising a noise reduction filter (20), an adaptive predictive filter (22) and an interpolator (24), connected in series, wherein the fundamental extractor outputs a fundamental signal; a subtractor (13) for calculating a difference between the signal source and the fundamental signal. A corresponding method is also presented.

METHOD AND APPARATUS FOR HARMONIC DETECTION**FIELD OF INVENTION**

The present invention relates generally to harmonics detection, and more particularly to reducing latency in
5 harmonics detection.

BACKGROUND

In electrical motors, it is known to detect harmonics e.g. using a measurement of total harmonic distortion.

It is known to use filters to obtain the fundamental
10 frequency of the signal source, whereby the fundamental frequency can be subtracted from the signal source to obtain only the harmonic distortions. When the harmonic distortion is sufficiently large, error handling can be automatically triggered.

15 However, such filters introduce delays, which can prolong the time until critical error handling is started.

SUMMARY

It is therefore an object of the present invention to reduce the latency introduced when measuring harmonic
20 distortion.

A first aspect of the invention is a harmonics detector for detecting harmonics of a signal source. The harmonics detector comprises: a fundamental signal extractor comprising a noise reduction filter, an adaptive
25 predictive filter and an interpolator, connected in series, wherein the fundamental extractor outputs a fundamental signal; a subtractor for calculating a

difference between the signal source and the fundamental signal.

The harmonics detector may be adapted to detect harmonics of rotating electrical machines.

- 5 The noise reduction filter may be a median filter, wherein the median filter is arranged to provide as its output a sample representing a median value of a plurality of most recent samples of the signal source.

10 The adaptive predictive filter may predict two sample steps ahead to compensate for the delay of the median filter and the interpolator.

The adaptive predictive filter may comprise a first and a second finite impulse response filter, where the first finite impulse response filter generates the output of
15 the adaptive predictive filter based on the input to the adaptive predictive filter and coefficients calculated by the second finite impulse response filter.

The coefficients to the first finite impulse response filter may be updated using a least mean square
20 algorithm.

The coefficients of the first filter may be updated according to the following formula:

$$H(n+1) = H(n) + 2 \times u \times e(n) \times X(n-2)$$

where:

25 $e(n) = x(n) - H^T(n) \times X(n-2)$ is the error,
 $H(n) = [h(0), \dots, h(n-1)]^T$ is the coefficient vector, and

$X(n-2) = [x(n-2), \dots, x(n-N-1)]^T$ is the data vector.

A second aspect of the invention is a method, performed in a harmonics detector, for detecting harmonics of a signal source. The method comprises the steps of:

reducing noise of the signal source, predicting a future
5 sample with an adaptive predictive filter, performing an interpolation, to thereby obtain a fundamental signal, and calculating a difference between the signal source and the fundamental signal to obtain a harmonics component of the signal source.

10 The method may further comprise the step of calculating a root mean square of the harmonics signal.

It is to be noted that any feature of the first and the second aspects may, where appropriate, be applied to any other aspect.

15 With the solution presented above, the latency is greatly reduced for harmonics calculations of signals. This allows for a quicker response to potential problems which can e.g. automatically respond to any fault condition by taking appropriate action. Furthermore, the resulting
20 measurement can be used to estimate losses e.g. in rotating electrical machines. With the adaptive filter, frequency variations can be followed, e.g. due to slip variations in asynchronous rotating electrical machines, or due to variations in a frequency of a power supply.

25 Generally, all terms used in the claims are to be interpreted according to their ordinary meaning in the technical field, unless explicitly defined otherwise herein. All references to "a/an/the element, apparatus, component, means, step, etc." are to be interpreted
30 openly as referring to at least one instance of the

element, apparatus, component, means, step, etc., unless explicitly stated otherwise. The steps of any method disclosed herein do not have to be performed in the exact order disclosed, unless explicitly stated.

5 BRIEF DESCRIPTION OF DRAWINGS

The invention is now described, by way of example, with reference to the accompanying drawings, in which:

Fig 1 is a schematic diagram showing an environment where the present invention can be applied,

10 Fig 2 is a schematic block diagram of the predictive filter of Fig 1,

Fig 3 comprises two graphs illustrating an extracted fundamental signal according to the predictive filter of Fig 2,

15 Fig 4 is a schematic block diagram illustrating the adaptive predictor of Fig 2,

Fig 5 comprises two graphs illustrating an adaption time of the predictive filter of Fig 1, and

20 Fig 6 is a flow chart illustrating a method of harmonic detection in the harmonics detector of Fig 1.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

The present invention will now be described more fully hereinafter with reference to the accompanying drawings, in which certain embodiments of the invention are shown.

25 This invention may, however, be embodied in many different forms and should not be construed as limited to the embodiments set forth herein; rather, these

embodiments are provided by way of example so that this disclosure will be thorough and complete, and will fully convey the scope of the invention to those skilled in the art. Like numbers refer to like elements throughout the
5 description.

Fig 1 is a schematic diagram showing an embodiment of a harmonics detector 1. An input signal 8 is fed to the harmonics detector 1. The harmonics detector can be detected using digital filters, e.g. one or more of using
10 digital signal processor (DSP), field programmable gate array (FPGA), central processing unit (CPU), application-specific integrated circuit (ASIC).

The input signal 8 is a signal with a fundamental signal and potential harmonics components that are to be
15 detected. For example, the input signal can be a detected current or voltage relating to a rotating electrical machine 6, such as an electrical motor. The signal 8 is an analogue signal. It is to be noted that the signal can be from other signal sources where it is useful to
20 extract harmonic components.

An A/D (analogue to digital) converter 10 converts the input signal 8 from analogue to digital format by sampling the input signal, thereby yielding a stream of quantified samples, i.e. a digital signal. The frequency
25 for the sampling is selected appropriately, e.g. between 5 - 50 kHz.

The digital signal is fed to a fundamental signal extractor 12 and a subtractor 13. As will be shown in more detail below, the predictive filter provides the
30 fundamental signal 9 of the input signal, without

producing a delay. Consequently, the subtractor 13 can find any difference between the incoming signal and the fundamental signal 9 for detecting harmonic components. The output of the subtractor is thus the harmonic components of the input signal 8 without any delay introduced by the filter 12. An RMS (root mean square) calculator 14 then calculates a root mean square value of the harmonics components to obtain a distortion measurement. It is to be noted that even if the fundamental signal extractor 12 is not perfect in extracting the fundamental signal 9 of the input signal 8, it is still usable to detect harmonic components.

Fig 2 is a schematic block diagram of the fundamental signal extractor 12 of Fig 1. The fundamental signal extractor 12 has three blocks. The first block is a noise reduction filter 20. In this embodiment, the noise reduction filter is a three point median filter 20, which removes disturbing impulses. The median filter 20 is a non-linear filter that operates e.g. by sorting the samples inside the moving filter window by magnitude, choosing the middle value, thereby removing completely isolated impulses. Instead of a three point median filter, any other appropriate median filter could be used, e.g. a five point or seven point filter.

The next component is an adaptive predictor 22, which predicts two steps ahead, one to compensate the delay in the median filter and the other to allow interpolation in the last stage. This filter behaves as a band pass filter with a narrow pass-band centred at the main frequency. Since an adaptive predictor 22 is used, the prediction step remains accurate, even if the line frequency changes over time. Thus, the filter is also able to work in much

higher or lower frequencies, only by changing some control parameters, such as the sampling frequency. For small variations in the frequency, no adjustments need to be made. The frequency range is limited only by the

5 Nyquist frequency of the discrete time system. An interpolator 24 interpolates the signal to get a continuous signal. The interpolator can be any suitable interpolator, e.g. a polynomial interpolator of a suitable degree. In one embodiment, the interpolator is a

10 sixth degree polynomial interpolator.

Fig 3 comprises two graphs illustrating an extracted fundamental signal according to the predictive filter of Fig 2. The upper graph 30 shows a corrupted input signal, i.e. an input signal with a significant harmonics

15 component, and the lower graph 32 shows the fundamental signal as output from the predictive filter of Fig 2.

Fig 4 is a schematic block diagram illustrating components of the adaptive predictor of Fig 2. There are two finite impulse response filters (FIR), a first FIR

20 filter 43 and a second FIR filter 44.

The second FIR filter 44 of the inside loop produces and updates 41 update coefficients for the first FIR filter 43. The coefficients of the adaptive filter are updated using the Least Mean Square algorithm (LMS), which is

25 selected due to its robustness and computational simplicity. The coefficient update formula is

$$H(n+1) = H(n) + 2 \cdot u \cdot e(n) \cdot X(n-2) \quad (1)$$

where

$$e(n) = x(n) - HT(n) \cdot X(n-2) \quad \text{is the error} \quad 50$$

$H(n) = [h(0), \dots, h(n-1)]^T$ is the coefficient vector 41

5 $X(n-2) = [x(n-2), \dots, x(n-N-1)]^T$ is the data vector 40
containing samples of the
input signal

The constant u is selected to be small enough to
guarantee the stability of the LMS algorithm. A two
10 sample delay element 42 is provided in the inner loop.
The adaptive filter configuration has been tested in
MATLAB/SIMULINK, with 22 coefficients. The filter shows
good convergence properties and a significant reduction
of the distortion of the primary sinusoidal component.

15 Fig 5 comprises two graphs illustrating an adaption time
of the predictive filter of Fig 1. The upper graph 60
shows an exemplary input signal being an induction motor
current and the lower graph shows the output of the RMS
calculator 14 (Fig 1). In an initial period 64 of about
20 0.2s, the adaptive predictive filter is still adjusting
to the fundamental signal. Since the fundamental signal
has not yet been found, the difference between the input
signal and the fundamental signal of the filter is
relatively large. In the second period 66 shown, however,
25 the adaptive predictive filter 22 has adjusted to the
input signal and essentially only the harmonics
components of the input signal contribute to the RMS
value shown in the graph 62.

Fig 6 is a flow chart illustrating a method of harmonic
30 detection in the harmonics detector of Fig 1.

In an initial reduce noise step 70, noise is reduced of the input signal, e.g. using the median filter 20 of Fig 2.

5 In a predict future samples step 72, future samples of the fundamental signal are predicted, e.g. using the adaptive predictive filter 22 of Fig 2.

In a interpolate step 74, the signal is interpolated, e.g. using the interpolator 24 of Fig 2, yielding an fundamental signal.

10 In a calculate difference step 76, a difference between the input signal and the fundamental signal is calculated, yielding a harmonics component.

In an optional calculate RMS step 78, the root mean square of the difference is calculated, which is used to
15 obtain the distortion measurement. After this step, the method ends. Alternatively, the method is repeated after this step for continuous harmonics detection.

The invention has mainly been described above with reference to a few embodiments. However, as is readily
20 appreciated by a person skilled in the art, other embodiments than the ones disclosed above are equally possible within the scope of the invention, as defined by the appended patent claims.

CLAIMS

1. A harmonics detector (1) for detecting harmonics of a signal source (8) comprising:

5 a fundamental signal extractor (12) comprising a noise reduction filter (20), an adaptive predictive filter (22) and an interpolator (24), connected in series, wherein the fundamental extractor (12) outputs a fundamental signal;

10 a subtractor (13) for calculating a difference between the signal source (8) and the fundamental signal (9).

2. The harmonics detector (1) according to claim 1, wherein the harmonics detector (1) is adapted to detect harmonics of rotating electrical machines (6).

15 3. The harmonics detector (1) according to claim 1 or 2, wherein the noise reduction filter (20) is a median filter, wherein the median filter is arranged to provide as its output a sample representing a median value of a plurality of most recent samples of the signal source.

20 4. The harmonics detector (1) according to any one of the preceding claims, wherein the adaptive predictive filter predicts two sample steps ahead to compensate for the delay of the noise reduction filter (20) and the interpolator (24).

25 5. The harmonics detector (1) according to any one of the preceding claims, wherein the adaptive predictive filter (22) comprises a first and a second finite impulse response filter (43, 44), where the first finite impulse response filter (43) generates the output of the adaptive
30 predictive filter (22) based on the input to the adaptive

predictive filter (22) and coefficients calculated by the second finite impulse response filter (44).

6. The harmonics detector (1) according to claim 5, wherein the coefficients to the first finite impulse response filter (43) are updated using a least mean square algorithm.

7. The harmonics detector (1) according to claim 5 or 6, wherein the coefficients of the first filter are updated according to the following formula:

$$H(n+1) = H(n) + 2 \times u \times e(n) \times X(n-2)$$

where:

$e(n) = x(n) - H^T(n) \times X(n-2)$ is the error,

$H(n) = [h(0), \dots, h(n-1)]^T$ is the coefficient vector, and

$X(n-2) = [x(n-2), \dots, x(n-N-1)]^T$ is the data vector.

8. A method for detecting harmonics of a signal source, the method comprising the steps, performed in a harmonics detector, of:

reducing (70) noise of the signal source,
predicting (72) a future sample with an adaptive predictive filter,
performing an interpolation (74), to thereby obtain a fundamental signal, and
calculating (76) a difference between the signal source and the fundamental signal to obtain a harmonics component of the signal source.

9. The method according to claim 8, further comprising the step of calculating (78) a root mean square of the harmonics signal.

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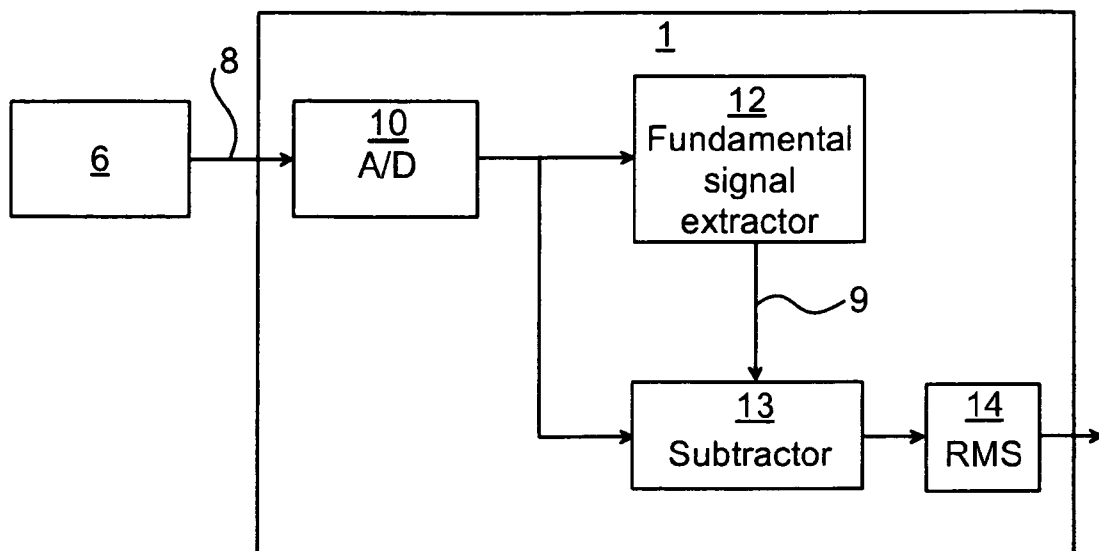


Fig. 1

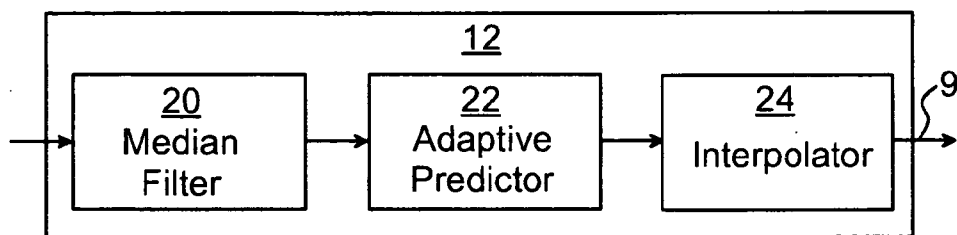


Fig. 2

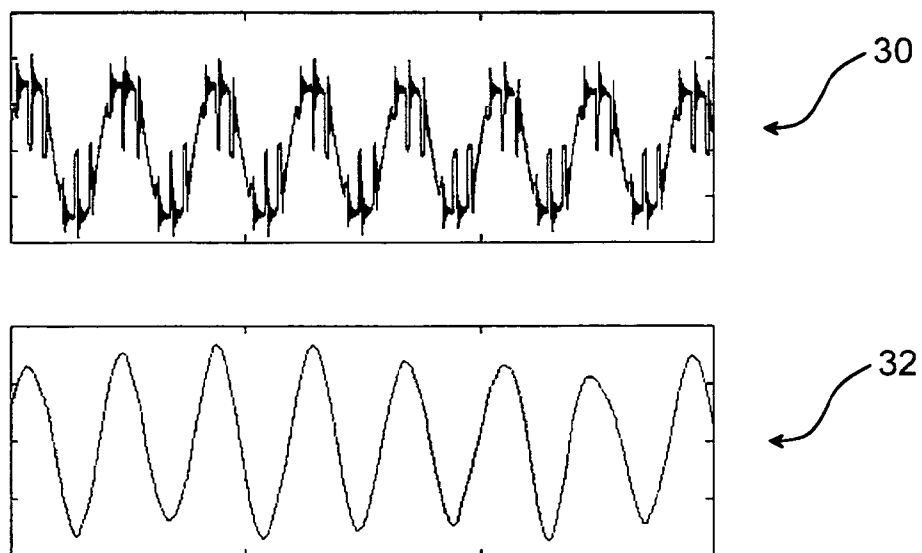


Fig. 3

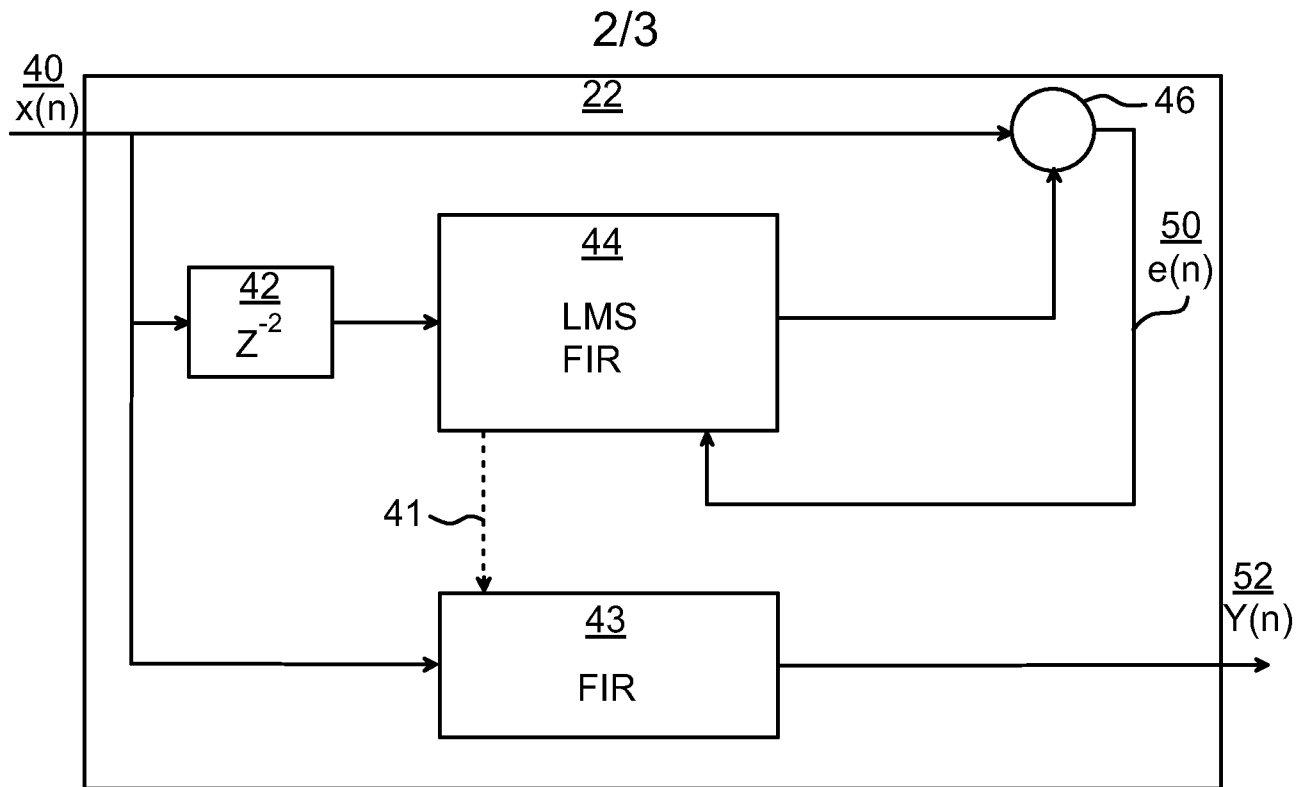


Fig. 4

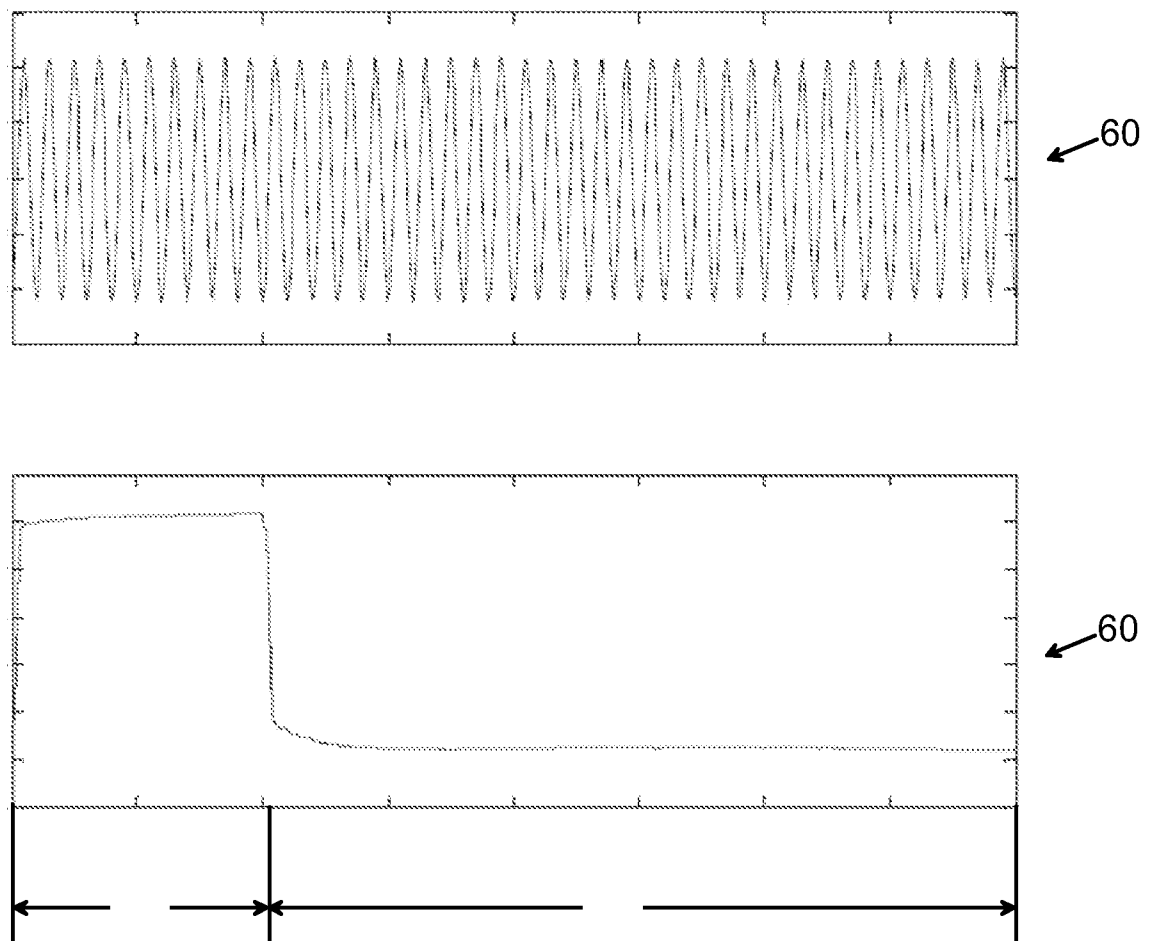


Fig. 5

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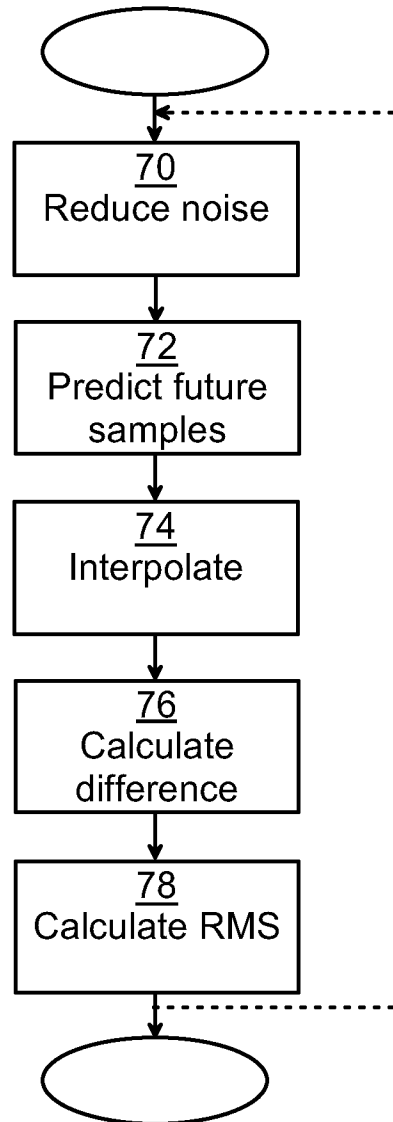


Fig. 6

INTERNATIONAL SEARCH REPORT

International application No
PCT/EP2009/058864

A. CLASSIFICATION OF SUBJECT MATTER
INV. G01R31/34 G06F17/14
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)
G01R G06F

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)

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	<p>OLLI VAINIO ET AL: "Multistage Adaptive Filters for In-Phase Processing of Line-Frequency Signals" IEEE TRANSACTIONS ON INDUSTRIAL ELECTRONICS, IEEE SERVICE CENTER, PISCATAWAY, NJ, USA, vol. 44, no. 2, 1 April 1997 (1997-04-01), XP011023251 ISSN: 0278-0046 the whole document</p> <p style="text-align: center;">----- -/-</p>	1-9

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

☒ See patent family annex.

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INTERNATIONAL SEARCH REPORT

International application No

PCT/EP2009/058864

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	<p>VAINIO O ET AL: "Multistage adaptive filters for in-phase processing of line-frequency signals" INSTRUMENTATION AND MEASUREMENT TECHNOLOGY CONFERENCE, 1997. IMTC/97. PROCEEDINGS. SENSING, PROCESSING, NETWORKING., IEEE OTTAWA, ONT., CANADA 19-21 MAY 1997, NEW YORK, NY, USA, IEEE, US LNKD-DOI:10.1109/IMTC.1997.603986, vol. 1, 19 May 1997 (1997-05-19), pages 428-433, XP010233673 ISBN: 978-0-7803-3747-3 the whole document</p>	1-9
A	<p>US 5 629 870 A (FARAG SAMIR F [US] ET AL) 13 May 1997 (1997-05-13) abstract column 2, line 38 - column 3, line 52</p>	1-9
A	<p>Noah P. Bethel: "Identifying Motor Defects Through Fault Zone Analysis" 30 August 2004 (2004-08-30), XP002577661 Retrieved from the Internet: URL: http://www.maintenanceworld.com/Articles/pdma/Identifying-Motor-Defects.htm [retrieved on 2010-04-14] page 1 - page 2; figures 1,2</p>	1-9

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/EP2009/058864

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US 5629870	A	13-05-1997	NONE