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Datasheet for the decision
of 12 October 2016

Case Number: T 1525/11 - 3.5.07

Application Number: 05719929.1

Publication Number: 1724928

IPC: H03M1/12, H03F3/45, H04L29/02

Language of the proceedings: EN

Title of invention:
Signal processing device and method, signal processing program, and recording medium where the program is recorded

Applicant:
Japan Science and Technology Agency

Headword:
Fluency A/D functions/JAPAN SCIENCE AND TECHNOLOGY AGENCY

Relevant legal provisions:
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Decisions cited:

Catchword:

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Case Number: T 1525/11 - 3.5.07

D E C I S I O N
of Technical Board of Appeal 3.5.07
of 12 October 2016

Appellant: (Applicant)	Japan Science and Technology Agency 4-1-8-, Honcho Kawaguchi-shi, Saitama 332-0012 (JP)
Representative:	Beetz & Partner mbB Patentanwälte Steinsdorfstraße 10 80538 München (DE)
Decision under appeal:	**Decision of the Examining Division of the European Patent Office posted on 19 January 2011 refusing European patent application No. 05719929.1 pursuant to Article 97(2) EPC.**

Composition of the Board:

Chairman	R. Moufang
Members:	M. Rognoni
	R. de Man

Summary of Facts and Submissions

I. The applicant (appellant) appealed against the decision of the Examining Division to refuse European patent application no. 05719929.1.

II. In the decision under appeal, the Examining Division held, *inter alia*, that claim 1 of the sole request submitted during the oral proceedings on 17 November 2010 was not clear within the meaning of Article 84 EPC, and that its subject-matter did not involve an inventive step (Article 56 EPC) in view of the following prior art:

D2: M. Obata *et al.*: "An approximation of data points by piecewise polynomial functions and their dual orthogonal functions", Signal Processing, Elsevier Science Publishers, B.V. Amsterdam NL, Vol. 80, no. 3, 1 March 2000, pages 507 - 514.

III. With the statement of grounds of appeal, the appellant filed a set of claims 1 to 6, and requested that the decision of the Examining Division be set aside and a patent be granted on the basis of the newly submitted claims.

IV. The appellant was summoned to oral proceedings to be held on 12 October 2016.

V. In a communication pursuant to Article 15(1) RPBA, the Board raised objections under Articles 84 and 56 EPC.

VI. In reply to the Board's communication, the appellant filed, with letter dated 5 October 2016, a new set of claims 1 to 6.

VII. At the end of the oral proceedings, which were held as scheduled on 12 October 2016, the Chairman pronounced the Board's decision.

VIII. The appellant requested that the decision under appeal be set aside and a patent be granted on the basis of claims 1 to 6 filed with the letter dated 5 October 2016.

IX. Claim 1 of the appellant's sole request reads as follows:

"A signal processing method comprising the steps of:

providing a plurality of fluency AD functions, which are sampling functions for acquiring discrete signals from continuous waveform signals to obtain a function value of the fluency sampling functions at sampling points, and further an inner product operation of obtained discrete signals, and the sampled value of the input signal is executed based on the fluency information theory and categorized with parameters m, where m is a positive integer from 1 to ∞, denoting that the fluency AD function is only continuously differentiable as often as (m-2) times,

sampling the continuous waveform signal which is inputted to get sampling values; the acquired discrete signals being used to provide approximate sampling data of an interpolation range by using each of the fluency functions from which one is selected which minimizes an approximation error,

characterized by

finding, within a certain period, inner product operating values by executing inner product operations between values of the inputted continuous waveform signal and the plurality of fluency AD functions having each different parameters m;

judging the differences between the sampling values and the inner product operating values at the sampling points;

determining the parameter m linked to the period for calculating the inner product, in which values of the differences come to minimum; and

outputting a combination of a parameter m signal indicating said determined parameter m and a discrete signal composed of a string of the sampling values or a combination of a parameter m signal indicating the parameter m and a discrete signal composed of a string of said inner product operating values as an output signal."

Claims 2 to 5 are dependent on claim 1.

Claim 6 reads as follows:

"A signal processing device comprising:

a plurality of fluency AD function generators (3) which provide a plurality of fluency AD functions which are sampling functions for acquiring discrete signals from continuous waveform signals to obtain a function value of the fluency sampling functions at sampling points, and further an inner product operation of obtained discrete signals, and the sampled value of the input signal is executed based on the fluency information theory and categorized with parameter m, where m is a positive integer from 1 to ∞, denoting that the fluency function is only continuously differentiable as often as (m-2) times;

a sampling circuit (2) which samples the continuous waveform signal which is inputted to get sampling values, the acquired discrete signals being

used to provide approximate sampling data of an interpolation range by using each of the fluency functions from which one is selected which minimizes an approximation error;
characterized by

a plurality of operating units (4) which output inner product operating values by executing inner product operations between the values of the inputted continuous waveform signal and the plurality of fluency AD functions having each different parameters m;

a judging unit (8) which judges differences between the sampling values and the inner product operating values at the sampling points, outputted from the operating unit (4), and determines the parameter m in which values of the differences come to minimum; and

an output device (9) which outputs a combination of a parameter m signal indicating the predetermined parameter m and a discrete signal composed of a string of the sampling values or a combination of a parameter m signal indicating the predetermined parameter m and a discrete signal composed of a string of the inner product operating values as an output signal."

X. The appellant's arguments relevant to the Board's decision may be summarised as followed:

The present invention was characterised by providing a plurality of fluency analogue-to-digital (A/D) functions as sampling function for acquiring discrete signals from continuous waveform signals. As known from the fluency information theory, fluency functions were categorised by a parameter m, where m was a positive integer from 1 to ∞. The value of the parameter m of the fluency A/D function to be used for the digital conversion of a continuous waveform signal was determined on the basis of the differences between values of the input waveform and values obtained by executing inner product operations between the input waveform signal and a plurality of fluency A/D functions. The parameter m of the A/D function which minimised these differences was then selected and combined with a string of sampling values of the input waveform signal to provide an output signal representing the A/D converted input signal.

An essential contribution of the present invention to the state of the art was the combination of a parameter m signal, which indicated the selected parameter m, and of a discrete signal composed of a string of sampling values of the input signal. The remaining features recited in claim 1 were based on the fluency information theory and did not require further elucidation, as the fluency information theory had been developed long before the priority date of the present application and was thus known to the skilled person. As the claims were clear to the skilled person, they complied with Article 84 EPC.

Document D2 disclosed the following technologies:

- Obtaining a discrete data string by sampling the input data.
- Thinning out the discrete data string by a predetermined interval.
- Executing interpolation operations to the thinned-out data by applying plural D/A functions and judging the error between the values of the interpolation operation and the discrete data string to determine an A/D function. The best downsampling basis (A/D function) that minimised a function representative of this error could be obtained from the upsampling basis (D/A function).

Document D2 did not teach to determine the optimum parameter m for the A/D function. In fact, this document used D/A functions to determine the appropriate A/D function, while according to the present invention, a plurality of A/D functions was used to identify the parameter m of the A/D function to be used for analogue-to-digital conversion.

Even if document D2 provided some theoretical background to A/D and D/A conversion based on fluency functions, it did not disclose its practical implementation. On the other hand, the present application developed the theoretical approach of document D2 and proposed an industrially feasible device, which involved an inventive step with respect to document D2. Hence, the subject-matter of claim 1 complied with Article 56 EPC and provided a basis for the grant of a patent.

Reasons for the Decision

1. The appeal is admissible.

2. The present application *"relates to a signal processing device and a signal processing method for generating discrete signals by using sampling from signals which change in time such as video (moving picture), image, and audio signals or those used for measurement and control"* (application as filed, page 1, lines 7 to 11).

   The addressed problem consists essentially in determining the functions best suited for the analogue-to-digital conversion of a continuous signal and for reproducing the original analogue signal from the digital samples.

2.1 The theoretical background of the invention is provided by the so-called *"fluency information theory"* according to which all *"signal properties*" can be classified by a *"fluency function*" having a parameter m which indicates that the function is *"continuously differentiable only as often as (m-2) times"* (*ibid.* page 4, lines 11 to 13 and 20 to 22).

In particular, "*a fluency DA function of fluency functions is given a numeric value at the k-th targeted sampling point kτ, where τ is a sampling interval. The fluency DA function becomes 0 at the other sampling points"* (*ibid.* page 4, lines 16 to 19).

As the parameter m identifies a particular *"fluency function*", the problem referred to above is essentially solved by determining the parameter m.

2.2 As explained in the description (page 8, line 26 to page 9, line 7), the "sampling function" A/D and the "inverse sampling function" D/A are orthogonal and expressed by means of the same parameter m.

2.3 A signal processing device according to the invention (*ibid.* page 19, line 7 to 24), *"uses the sampling function to acquire <u>a discrete signal</u> from a continuous waveform signal based on the fluency information theory. The embodiment aims at videos and images, and parameter m is set to three types 2, 3, and ∞. This is because an analysis result shows that three parameters m = 2, 3, and ∞ cover almost all signal properties of signals acquired from videos and images. The invention is not limited to these three parameters. Obviously, it may be preferable to choose four parameters, i.e., m=1, 2, 3, and ∞, for example, when diagrams are also included. According to the embodiment, the digital* *signal processing generates a discrete signal from a continuous waveform signal. For this reason, an analog input signal is once sampled at an interval sufficiently shorter than sampling interval τ and then is PCM encoded. Further, the sampling function with m = 2 or 3 is settled within the finite span 0 to (J-1)τ, where J is the number of sampling points and (J-1)τ is the length. An inner product is also taken for each sampling point within this range*" (underlining added).

The appellant's request

3. Claim 1 of the appellant's request relates to a "*signal processing method*" comprising the following steps itemised by the Board:

(a) providing a plurality of fluency AD functions,
 (i) which are sampling functions for acquiring discrete signals from continuous waveform signals
 (ii) to obtain a function value of the fluency sampling functions at sampling points, and further an inner product operation of obtained discrete signals, and
 (iii) the sampled value of the input signal is executed based on the fluency information theory and
 (iv) categorized with parameters m,
 (v) where m is a positive integer from 1 to ∞,
 (vi) denoting that the fluency AD function is only continuously differentiable as often as (m-2) times;

(b) sampling the continuous waveform signal which is inputted to get sampling values;

(i) the acquired discrete signals being used to provide approximate sampling data of an interpolation range by using each of the fluency functions from which one is selected which minimizes an approximation error,

(c) finding, within a certain period, inner product operating values by executing inner product operations between values of the inputted continuous waveform signal and the plurality of fluency AD functions having each different parameters m;

(d) judging the differences between the sampling values and the inner product operating values at the sampling points;

(e) determining the parameter m linked to the period for calculating the inner product, in which values of the differences come to minimum; and

(f) outputting

(i) a combination of a parameter m signal indicating said determined parameter m and a discrete signal composed of a string of the sampling values or

(ii) a combination of a parameter m signal indicating the parameter m and a discrete signal composed of a string of said inner product operating values as an output signal.

Interpretation of claim 1

4. Claim 1 on file differs from claim 1 submitted with the statement of grounds of appeal essentially in that it further comprises features (a)(ii), (a)(iii) and (b)(i), and in that it specifies that the inner product operations (feature (c)) are executed *"within a certain period"*.

4.1 In the Board's opinion, these amendments relate to aspects inherent to the *"fluency information theory"* which is at the basis of the signal processing of the present invention. Hence, to the reader familiar with the fluency information theory, the subject-matter of the present claim 1 is essentially the same as the subject-matter of the previous claim 1.

4.2 Feature (a) specifies that there is a class of functions called *"fluency AD functions"* which are distinguished by a parameter m (where m is an integer from 1 to ∞) and can be used as sampling functions for sampling continuous waveform signals.

Step (b) relates to the sampling of the input signal to obtain discrete values which are used in step (d) and provide the output string of values according to step (f)(i) (cf. application as filed, page 20, lines 8 to 24).

Steps (c), (d) and (e) specify how to perform, over a *"certain period"*, the selection of the A/D function which provides the best approximation of the sampling values obtained in step (b).

Step (f) defines the results of the signal processing according to claim 1 as a combination of a parameter **m** (actually the parameter m of the A/D function selected for the *"certain period")* and a string of corresponding sampling values, or, alternatively, of inner product operating values.

4.3 At the oral proceedings, the appellant essentially argued that the fluency information theory, on which the signal processing according to the present invention was based, had been developed long before the filing date of the present invention and was generally known to the skilled person. Hence, there was no need to specify further aspects of the fluency theory in the claim or to identify the fluency AD functions further.

In the appellant's opinion, the gist of the invention consisted in representing a continuous waveform signal by a combination of a parameter m, which identified a class of fluency functions, and a string of sampling values. The combination of data obtained at the transmitting end of a communication system could then be used at the receiving end to determine the appropriate fluency D/A function and obtain from the string of sampling values a continuous waveform representative of the input signal.

4.4 The Board sees no reason to question the appellant's submissions relating to the technical knowledge generally available at the time of filing of the present application and accepts that the skilled person was sufficiently familiar with the fluency information theory to be aware of the properties of fluency functions and, in particular, of the fact that fluency AD functions and corresponding DA functions could be identified by means of their parameters m, where $m = 1$ to ∞.

4.5 Furthermore, the Board considers that, in the light of the fluency information theory, it is possible to understand the claimed invention and to evaluate its inventive merit.

Inventive step

5. Document D2 provides in its introduction some background information on the so-called *"fluency theory"*. In particular, D2 (page 508, left-hand column, first paragraph) points out that the "[t]*here are many types of analog signals (time waveforms, time series) which need to be processed for practical use: continuous signals, discontinuous signals, smooth (or differentiable) signals, and signals composed of one or more combinations of these. The fluency theory provides a generalized model for these various types of signals by introducing a powerful tool called the fluency* ***digital/analog (D/A) conversion functions*** *and the fluency* ***analog/digital (A/D) conversion functions****. The D/A and the A/D functions are piecewise polynomial functions that are not orthogonal to themselves but are orthogonal to each other"*.

5.1 Furthermore, in the second paragraph of the introduction (page 508, left-hand column), document D2 specifies that, "[i]*n the fluency model, signals are classified in terms of their continuous differentiability,* ***m****. If the target, namely an analog signal, belongs to class* ***m****, then it can be discretized by a class* ***m*** *A/D function at every time interval of* ***h****. This A/D converted signal will be equivalent to the target signal that is sampled every time* ***h*** *intervals, and can be perfectly reconstructed back using a class* ***m*** *D/A function. If the target does not belong to* ***m****, class* ***m*** *A/D and D/A functions work together to give a class* ***m*** *signal that best approximates the target in terms of* L_2 *- norm* [i.e. least squares]*".*

As pointed out in a note at the bottom of page 508 of D2 (left-hand column), the term "*A/D converted signal"* denotes a signal that is made discrete by the A/D functions to differentiate it from a "*sampled signal*".

5.2 Figure 1 of D2 shows an example of how a continuous waveform signal, which does not "belong to **m**", is *"A/D converted"* to generate a set of *"expansion coefficients"* (*i.e.* the *"discretized signal"*), and how the discretised signal is D/A converted into a signal which best approximates the original signal.

It is further stated in D2 (paragraph bridging the left-hand and right-hand columns of page 508) that *"if the class of the **A/D** and the **D/A** functions differ, the best approximation cannot be obtained. It should be noted, that once an analog signal is made discrete, there are no means to find out which A/D function class was used. Therefore, past applications of the fluency theory focused on interpolating a set of data points using a class **m D/A** function, under the assumption that the data points were discretized via class **m A/D** function".*

5.3 In summary, document D2 confirms that it was known to the skilled person to apply the fluency information theory to convert a generic signal into a *"discretized signal*" by means of an A/D conversion function identified, according to the fluency theory, by a parameter **m**, and that the best approximation of the original signal could be obtained by interpolating the *"discretised signal"* by means of a D/A conversion function belonging to the same class **m** of the A/D function.

5.4 In the Board's opinion, a straightforward conclusion that can be drawn from this background knowledge is that the best interpolation of a discretised signal resulting from the A/D conversion of an input signal can only be obtained when the values of the discretised signal and the corresponding parameter **m** are communicated to the interpolator so that the appropriate D/A function is selected.

5.5 Hence, it would be obvious to a skilled person, who wished to develop a method for processing a signal according to the fluency information theory and to ensure the best interpolation of the A/D converted signal, to output a combination of a parameter **m** signal and a discrete signal composed of a string of corresponding sampling values according to step (f) recited in claim 1 of the appellant's request.

6. Features (a) to (e) of claim 1 (see Board's itemisation) relate to the A/D conversion of the input signal and to the selection of the A/D function which can give the best approximation.

6.1 In particular, feature (a) specifies aspects of the fluency functions which, as acknowledged by the appellant, are known to the skilled person conversant with the fluency information theory.

6.2 Feature (b) relates to the selection of the A/D function which minimises "an approximation error".

6.2.1 As shown in Figure 1 of document D2 (see 5.2 above), the D/A interpolation of an A/D converted signal which does not belong to a class **m** produces an approximation of the original signal. It goes without saying that the quality of the approximation depends on the choice of the A/D conversion function and ultimately on the parameter **m**.

This is also clearly illustrated by Figure 5 of D2 which shows the original target (Figure 5(a)) and the squared errors between the target and the approximations obtained with fluency vectors belonging to different m classes (Figure 5(b) to (d)). This comparison demonstrates that, in the first 40 samples, the target is best approximated by an m = 3 vector pair, whereas in the last 60 samples m = ∞ outperforms the others.

6.2.2 It should be noted that in sections 3 ("Approximation of data points") and 4 ("Implementation") document D2 refers no longer to "fluency function", but to "fluency vectors", since it is concerned with the interpolation of a given set of data points. It is however evident that the general teaching which can be extracted from these sections of document D2 applies to fluency functions in general.

6.2.3 As stated in D2 (page 512, right-hand column, last paragraph), the results shown in Figure 5 suggest the effectiveness of employing different classes of fluency vectors (or functions). Consequently, approximation of the A/D and D/A conversion can be improved if the class of the fluency vectors (or functions) can be adaptively changed.

6.2.4 In other words, feature (b) simply expresses the known fact that an A/D conversion of a signal which minimises the approximation error between the discrete values of the input signal and corresponding discrete values of the converted signal presupposes the selection of the appropriate A/D function and consequently the determination of the best parameter **m**.

6.3 Features (c), (d) and (e) relate to the selection of the parameter **m** which identifies the A/D function best suited to convert a certain portion of the input signal.

6.3.1 In particular, feature (c) describes the A/D conversion of the input waveform signal by means of different A/D conversion functions. As known from the fluency information theory (cf. D2, page 512, section 4.1, last two paragraphs), this is done by executing inner product operations between values of the input waveform signal and the A/D functions.

6.3.2 According to features (d) and (e), the parameter **m** is determined on the basis of the difference between the sampling values and the inner product operating values at the sampling points. In fact, as explained in the description, the comparison between the input signal and the A/D converted signal is carried out at certain discrete points in time (see original application, page 20, line 8 to page 21, line 16).

6.3.3 The fact that the selection of the fluency function (*i.e* the parameter **m**) which guarantees the best approximation of the input signal is performed on the basis of differences between sampling values of the input signal and the inner product operating values at the sampling points may at first sight appear unusual, since *a priori* the values resulting from the application of a generic A/D conversion function to a signal are not directly comparable with values of the input signal. However, according to the fluency theory (see D2, page 509, section 2.3), the conversion of a signal which does not belong to a certain class **m** by means of an A/D fluency function generates a set of expansion coefficients **s** that approximates the target signal when expanded with the corresponding D/A fluency function, whereas all signals of a class **m** can be expressed in the form of linear combinations of a D/A function <u>with their expansion coefficients being the target signal itself</u> (D2, page 509, left-hand column, first paragraph).

6.3.4 In other words, the expansion coefficients obtained by converting a generic waveform signal with a fluency A/D function of class **m** (cf. "inner product operating values" referred to in step (f)) are discrete values of the target signal, *i.e.* of a signal belonging to the signal space **m** (see section 2.1 of D2), and thus represent approximations of the values of the input signal sampled at corresponding time intervals.

6.3.5 Hence, it follows from the fluency theory that differences between the sampling values of an input signal and the inner product operating values at the sampling points (see feature (d)) are indicative of how well the A/D and D/A conversion approximates the target signal and that the best A/D conversion is the one that minimises these differences.

7. The appellant has, *inter alia*, argued that document D2 disclosed a theoretical idea of A/D and D/A conversion based on the fluency function, but did not disclose an apparatus for practical use of the theoretical idea.

7.1 Furthermore, the appellant has pointed out (see statement of grounds, page 4, second paragraph) that, according to the present invention, the parameter **m** was determined on the basis of A/D functions, while document D2 used the D/A functions to determine the A/D function.

8. As to the first argument of the appellant, the Board notes that claim 1 relates to a method and not to a device "for practical use". In fact, the claim steps express in general terms the basic operations required for the A/D conversion of a signal on the basis of the fluency information theory referred to by the appellant.

8.1 As to the disclosure in D2, the Board acknowledges that D2 is not directly concerned with the A/D conversion of a generic continuous signal. In fact, document D2 aims, *inter alia*, at approximating a set of data points and explains that the problem in applying the fluency functions for this purpose is the difficulty in finding out the class of the A/D function used to obtain the given target. Thus, it is assumed that a class **m** D/A function is the expansion basis for the signal that best approximates the given data set. In the detailed example (section 3 of D2), which describes the relation between the D/A basis and the target data points in a matrix form, the D/A function, whose values are obtained at discrete time intervals, is defined as an *"upsampling vector".* Differences between values of the initial set of data and interpolated values based on the upsampling vector (D/A vector) are used to determine the optimal downsampling vector (A/D vector), namely the A/D basis which best approximates the given target data. As referred to above, Figure 5 illustrates an example of this approach and which class **m** gives the best approximation of the original signal.

8.2 In section 5, the authors of document D2 conclude that they have shown the effect of employing different classes of fluency vector pairs and that the class that yields the best approximation changes according to the local characteristics of the target signal. *"This implies,* [sic] *that if we are able to locally analyze the characteristics of the target and adaptively change the class of fluency vectors according to the obtained results, approximations can be improved significantly".*

8.3 In summary, the Board considers that document D2 discloses all the teaching that a skilled person would need to arrive at the combination of steps according to the method of claim 1 without engaging in any inventive activity.

9. Hence, the Board comes to the conclusion that, in the light of document D2 and of general knowledge common in the field of signal processing and of fluency information theory, it would have been obvious to a skilled person to arrive at the subject-matter of claim 1 of the appellant's request (Article 56 EPC).

9.1 The same objections apply to the device claim 6 which comprises features closely reflecting the steps recited in claim 1.

10. As the appellant's sole request cannot form the basis for the grant of a patent, the appeal has to be dismissed.

Order

For these reasons it is decided that:

The appeal is dismissed.

The Registrar: The Chairman:

I. Aperribay R. Moufang

Decision electronically authenticated