



- (51) **International Patent Classification:** Not classified
- (21) **International Application Number:** PCT/IB2014/059252
- (22) **International Filing Date:** 26 February 2014 (26.02.2014)
- (25) **Filing Language:** English
- (26) **Publication Language:** English
- (30) **Priority Data:** 61/774,003 7 March 2013 (07.03.2013) US
- (71) **Applicant:** KONINKLIJKE PHILIPS N.V. [NL/NL]; High Tech Campus 5, NL-5656 AE Eindhoven (NL).
- (72) **Inventors:** HUANG, Jinfeng; c/o High Tech Campus, Building 5, NL-5656 AE Eindhoven (NL). HESSEL, Henk Albert; c/o High Tech Campus, Building 5, NL-5656 AE Eindhoven (NL).
- (74) **Agents:** STEFFEN, Thomas et al.; High Tech Campus, Building 5, NL-5656 AE Eindhoven (NL).

- (81) **Designated States** (unless otherwise indicated, for every kind of national protection available): AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BN, BR, BW, BY, BZ, CA, CH, CL, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IR, IS, JP, KE, KG, KN, KP, KR,

KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PA, PE, PG, PH, PL, PT, QA, RO, RS, RU, RW, SA, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TH, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

- (84) **Designated States** (unless otherwise indicated, for every kind of regional protection available): ARIPO (BW, GH, GM, KE, LR, LS, MW, MZ, NA, RW, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, RU, TJ, TM), European (AL, AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MK, MT, NL, NO, PL, PT, RO, RS, SE, SI, SK, SM, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, KM, ML, MR, NE, SN, TD, TG).

Declarations under Rule 4.17:

- as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))
- as to the applicant's entitlement to claim the priority of the earlier application (Rule 4.17(iii))

Published:

- without international search report and to be republished upon receipt of that report (Rule 48.2(g))

(54) **Title:** ADAPTIVE UP-SAMPLING FOR ACCURATE REAL-TIME INTERPOLATIONS

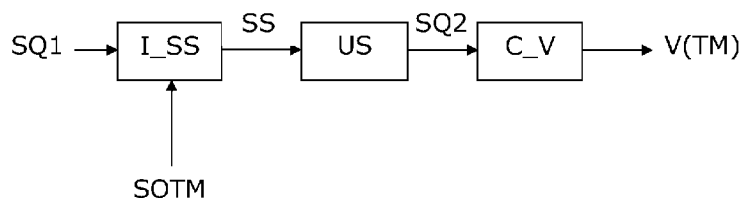


FIG. 2

(57) **Abstract:** A method and device for calculating a value representative of a signal at an interpolation time (SQTm), based on an input sequence (SQ1) representative of a temporal sampling of the signal. An adaptive up-sampling is performed by identifying a limited portion of the input sequence (SS), selected such as to temporally span the interpolation time (SQTm), thus allowing precise interpolation of input samples temporally local to the interpolation time (SQTm). The limited portion of the input sequence (SS), e.g. only one sample on each side of the interpolation time (SQTm), is then up-sampled to form an up-sampled sequence (SQ2), which has a rather low rate on average, due to the discarded input samples, e.g. equal to the sample rate of the input sequence (SQ1). However, locally around the interpolation time (SQTm), the sample rate is higher. Finally, it is possible to calculate a value representative of the signal at the interpolation time (V(TM)) in response to the up-sampled sequence (SQ2). Using linear FIR filters in the up-sampling process, a significant reduction of processing power can be saved, and still a high precision with a preserved phase information can be obtained, thus allowing the method to be used e.g. for optical shape sensing applications.



Adaptive up-sampling for accurate real-time interpolations

FIELD OF THE INVENTION

The present invention relates to the field of signal processing. More specifically, the invention relates to a method and a device with adaptive up-sampling for accurate real-time interpolations.

BACKGROUND OF THE INVENTION

An analog signal is a real-valued (or complex-valued) function $A(t)$, which is defined at every moment t in an (infinite/finite) interval. Its sampled signal at an A/D convertor is a discrete function $D(T_n)$ which has the same value as $A(T_n)$ at the discrete moments T_n without considering the quantization errors. However, all information about the signal between two adjacent samples is lost due to sampling. In some applications, the intermediate values are also needed, which are often interpolated based on discrete sampled points.

Up-sampling increases the sampling rate of a signal, which allows a more accurate interpolation. But at the same time it increases the amount of data to be processed and the computation requirements of the system to perform the computation. In real-time applications involving a large amount of data, it is often a challenge to achieve both high interpolation accuracy and in-time data processing.

Typically, the interpolation process introduces errors, which represent the deviations of the interpolation values from those of original signal at the same moments. In order to reduce the interpolation error, up-sampling is often applied, which increases the sampling rate of the signal. Generally speaking, the larger the sampling rate is, the smaller the interpolation error is. Apart from the interpolation error of the signal in the time domain, the interpolation also introduces frequency leakages in its frequency domain and phase errors which is crucial for some applications, e.g. in signal processing for real-time optical shape sensing.

SUMMARY OF THE INVENTION

It would be advantageous to provide a method and a device capable of providing an up-sampling of a sampled time signal in real-time with a high precision of phase representation of the signal, and still requiring a limited processing power.

In a first aspect, the invention provides a method for calculating a value representative of a signal at an interpolation time, based on an input sequence representative of the signal, the method comprising

- receiving the input sequence, wherein the input sequence comprises a series of input samples temporally sampled,
- receiving information about an interpolation time,
- identifying a limited portion of the input sequence, wherein the limited portion of the input sequence is selected in response to the interpolation time so as to temporally span the interpolation time,
- up-sampling the limited portion of the input sequence to an up-sampled sequence, and
- calculating a value representative of the signal at the interpolation time in response to the up-sampled sequence.

By ‘temporally span the interpolation time’ is understood that the limited portion of the input sequence at least includes one sample of the input sequence corresponding to a sample time lower than the interpolation time, and one sample of the input sequence corresponding to a sample time higher than the interpolation time, thus having samples representative of the signal value temporally at both sides of the interpolation time. By ‘interpolation time’ is understood interpolation moment or point in time, at which the precise value of the original signal represented in the input sequence is required.

The method is advantageous, since it is possible to skip processing one or more samples of the input sequence, thus saving processing power in the up-sampling process in an adaptive way in response to the interpolation time, i.e. the time where the precise value of the original signal is required. Especially, in case linear FIR filters are used as low pass filters to provide a correct phase representation, a significant amount of multiplications can be saved. Further, if the up-sampling rate is high, e.g. a factor of 16 or more, the method steps of identifying a limited sequence and up-sampling can be repeated, e.g. using an up-sampling factor of 2 or 4 in each up-sampling, and hereby the calculation power can be dramatically reduced. This allows real-time high precision applications running on a processor with limited processing power, e.g. real-time optical shape sensing.

In the following, a number of embodiments will be described.

In one embodiment, the limited portion of the input sequence can be selected such that it contains only input samples temporally near to the interpolation time. In one embodiment, the limited portion of the input sequence can be contains only input samples temporally adjacent to the interpolation time, i.e. only two samples, one sample on each side of the interpolation time is used. In one embodiment, the limited portion of the input sequence is selected such that it contains only a first plurality of input samples at times lower than the interpolation time and a second plurality of input samples at times higher than the interpolation time. Thus, especially 2, 3, 4, 5, 6 or more samples can be used on each side of the interpolation time, preferably the number is selected depending on the actual input sample rate, the interpolation rate, the further processing, choice of filters, and the up-sampling rate.

Such embodiments are advantageous for reducing the number of samples in the input sequence to a reduced number of samples which are then up-sampled. Only the samples temporally around the interpolation time are up-sampled, since only the samples of the input sequence which are relevant for precisely determining the value representing the original signal are taken into account in the up-sampling. Depending on the input sequence sample rate and the rate of incoming interpolation times, more or less computations can be saved.

The sample rate of the up-sampled sequence may especially be within 50-200% of a sample rate of the input sequence, on average. Especially the sample rate of the up-sampled sequence may be equal to or around equal to the sample rate of the input sequence. This is in spite the up-sampling rate that may be a factor of 2, 3, 4, 5, 6, 8 or more of the sample rate of the input sample, locally around the interpolation time, but due to the fact that only a limited number of the input samples are used to form the up-sampled sequence.

The method may comprise identifying a limited portion of the up-sampled sequence, wherein the limited portion of the up-sampled sequence is selected in response to the interpolation time so as to temporally span the interpolation time, and up-sampling the limited portion of the up-sampled sequence to a second up-sampled sequence, and calculating a value representative of the signal at the interpolation time in response to the second up-sampled sequence.

The up-sampling process may comprise convoluting with a Finite Impulse Response (FIR) low pass filter. For instance, if a precise phase representation is required, the Finite Impulse Response filter is a linear FIR filter.

For some applications, the information about interpolation time may be received at an interpolation rate, e.g. a fixed incoming rate per time. The interpolation rate may be lower than a sample rate of the input sequence on average over a sufficient long period, e.g. the interpolation rate may be 1-25% of the sample rate of the input sequence.

In one embodiment, the method comprises receiving information about a plurality of interpolation times, and identifying a limited portion of the input sequence, wherein the limited portion of the input sequence is selected in response to the plurality of interpolation times so as to form groups of the input samples temporally spanning the respective plurality of interpolation times, up-sampling the limited portion of the input sequence to an up-sampled sequence, and calculating values representative of the signal at the plurality of interpolation times in response to the up-sampled sequence.

The step of calculating the value representative of the signal at the interpolation time may comprise applying an interpolation algorithm to the up-sampled sequence, e.g. an interpolation algorithm comprising linear interpolation or other types of interpolation as known in the art.

It is to be understood that the up-sampling process can be performed in software, in hardware (by means of an up-sampling analog-to-digital converter), or a mix of both software and hardware, in case the up-sampling process is split into two or more stages.

In a second aspect, the invention provides a device comprising a processor arranged to receive an input sequence comprising a series of input samples temporally sampled, to receive information about an interpolation time, to identify a limited portion of the input sequence, wherein the limited portion of the input sequence is selected in response to the interpolation time so as to temporally span the interpolation time, to up-sample the limited portion of the input sequence to an up-sampled sequence, and to calculate a value representative of the signal at the interpolation time in response to the up-sampled sequence.

Especially, the device may comprises an optical shape sensor, wherein the signal is indicative of a shape of an optical shape sensor, and wherein the device is arranged to update a measure of shape of the optical shape sensor in response to the calculated value representative of the signal at the interpolation time. Such device requires a high precision, where phase representation is important for calculating the physical shape of the optical shape sensor in real-time.

The processor may be a single processor, such as a FPGA or a microprocessor, or a system comprising two or more single processors. Further, some of the processing may

be performed in hardware, while other parts of the processing are performed with a processor suitably programmed with software code.

Especially, the device may be a medical device, e.g. a medical device comprising an optical shape sensor.

It is appreciated that the same advantages and embodiments of the first aspect apply as well for the second aspect. In general the first and second aspects may be combined and coupled in any way possible within the scope of the invention. These and other aspects, features and/or advantages of the invention will be apparent from and elucidated with reference to the embodiments described hereinafter.

BRIEF DESCRIPTION OF THE DRAWINGS

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

FIG. 1 illustrates up-sampling according to prior art,

FIG. 2 illustrates a block diagram of basic parts of an embodiment,

FIGs. 3a and 3b illustrate a 2 times up-sampling embodiment,

FIGs. 4a and 4b illustrate FIR filters for use as low-pass filters in the up-sampling process.

FIGs. 5a and 5b illustrate a two-step up-sampling embodiment,

FIG. 6 illustrates the FIR filter used in the embodiment of FIGs. 5a and 5b,

FIG. 7 illustrates another two-step embodiment,

FIG. 8 illustrates a block diagram of an optical shape sensing embodiment,

FIG. 9 illustrates an example of optical elements of an optical shape sensing embodiment, and

FIG. 10 illustrates steps of a method embodiment.

DESCRIPTION OF EMBODIMENTS

FIG. 1 illustrates an example of up-sampling according to prior art. An incoming signal at sample rate f_s is zero inserted ZI (zero padding), and subsequently up-sampled by a factor of 4 and then filtered by low-pass filter LPF. A second step of 4 times up-sampling can follow, thus finally resulting in an output signal with a sample rate of $16 \cdot f_s$. Using a linear FIR low-pass filter LPF designed to preserve correct phase information, which is important e.g. for optical shape sensing applications, will result in an unacceptable large

number of multiplications. Thereby, leaving such up-sampling impossible for many real-time applications, if high precision and correct phase representation is required.

FIG. 2 shows a simple block diagram of an embodiment. An input sequence SQ1 of input samples is received. SQ1 may be a digital representation of input sample values representing values of an analog signal temporally sampled at a sample rate of e.g. 5 MHz. Further, a sequence of interpolation times SQTm is received at an incoming rate, e.g. at a rate of 2.5 MHz on average, at least preferably lower than the rate of SQ1. The method comprises identifying I_SS a sub sequence SS of the input sequence SQ1 adaptively to the sequence of interpolation times SQTm. Thus, only a limited number of samples of the input sequence SQ1 are selected to form the sub sequence SS, and the limited number of samples are selected such that only samples are included which are relevant for calculation of a precise signal value at the interpolation times SQTm. Thereby, some samples of the input sequence SQ1 can be discarded for further calculations. Only the selected sub sequence SS is then up-sampled US e.g. 2, 3, 4, 5, 8, 16 times or more to form an up-sampled sequence SQ2. However, note that the average sample rate of SQ2 may be decreased to a rate lower than the sample rate of the input sequence SQ1, or even lower than the incoming interpolation time SQTm rate, even though locally around the interpolation time, the sample rate is higher since here determined by the up-sampling rate. The up-sampled sequence SQ2 is then applied to a calculation algorithm C_V which calculates a value V(TM) representative of the original signal at the interpolation times. It is to be understood that the up-sampling process US can be performed in various ways as known to the skilled person, e.g. zero insertion (padding) and subsequent low pass filtering, preferably with a linear FIR filter to preserve correct phase information. It is to be understood that the input sequence SQ1 could be an up-sampled sequence in itself, e.g. an output of an up-sampling analog-to-digital converter, and in such case the step of identifying the sub sequence SS preferably comprises a low pass filter, e.g. a linear FIR filter, with a stop band of no higher than half of the rate of SQTm.

FIGs. 3a and 3b illustrate signal examples of an embodiment of the adaptive up-sampling according to the invention, where T is a time axis. In Fig. 3a, an input signal arrives in the form of samples at sample moments s1-s5 at an incoming sample rate. The value of the input signal is indicated vertically and illustrated as a dashed curve. The interpolation moments I1, I2 where the precise value of the input signal is required are indicated below. A 2 times up-sampling is performed on the input signal, but where the up-sampling is temporally determined by a disabling/enabling DE_E in response to the interpolation moments I1, I2, such that only a part of the input signal is up-sampled, namely

only input samples around the interpolation moments I1, I2. Thus, the factor 2 up-sampling procedure only involves introducing up-sampling moments u1, u2, and not between s2 and s3 and between s3 and s4, which would normally be done according to prior art up-sampling schemes. The method does not increase the sampling rate all the time. It is only enabled to increase the sampling rate around interpolation moments I1, I2. Thus, as seen to the right in Fig. 3a, the sample rate is only increased between sample s1 and s2, and between sample s4 and s5, where u1 and u2 are calculated by a low-pass filter (not shown).

In case of using linear interpolation LNP at the final step in Fig. 3b of determining the input value at the interpolation moments, the interpolation point at I1 is interpolated between the up-sampled point at u1 and the sample point at s2. In this case, it is also possible to skip the calculation of the irrelevant points during the up-sampling process, e.g. the point between s2 and s3. In this way it is possible to calculate the input signal value at the interpolation moments I1, I2 with high precision with a significantly reduced number of multiplications, thus being suitable for real-time applications.

Assume a sinusoidal input signal with a frequency of 1.25 MHz and the sampling frequency is at 5 MHz. For simplicity, an equal time distance for interpolation is assumed, even though this is not the case for many applications. The interpolation step is 1.01 times of the sampling step, i.e. the interpolation moment rate is lower than the input signal sample rate, namely 4.95 MHz (5 MHz/1.01). If up-sampling is carried out by two cascaded filters with an up-sampling factor 4 for each, sampled input data go through two cascaded up-sampling processes, which results in a 16 times data rate of the resulting up-sampled input signal, i.e. a sample rate of $16 \times 5 \text{ MHz} = 80 \text{ MHz}$ according to a prior art scheme. However, according to an embodiment of the invention, the sampled input signal first goes through a non-adaptive 4 times up-sampling process, which results in an output rate of 20 MHz. In the second up-sampling process, it has the same low pass filter as in the prior art scheme just described. However, only a part of the input signal sample close to the interpolation moments is filtered. Since the interpolation rate is about 4.95 MHz, the output of the adaptive up-sampling filter is about 19.8 MHz which is far less than the 80 MHz of the prior art scheme just described. This example shows that an adaptive up-sampling scheme can achieve a 16 times up-sampling accuracy with a 4 times up-sampling data rate. In the example, the adaptive up-sampling needs only about 25% computational requirements of the non-adaptive up-sampling in the second up-sampling process.

Further, the frequency leakage of the interpolation results based on 4 times and 16 times up-sampling signals can be significantly improved with the proposed method. A suppression of spurious frequency components can be suppressed by about 30 dB.

FIGs. 4a and 4b illustrate the low-pass filters which are used for the above example. The 49 tap filter in FIG. 4a is used for the first up-sampling process and the 33 tap filter in FIG. 4b is used for the second up-sampling process.

FIGs. 5a and 5b illustrate signal processing in the same manner as in FIGs. 3a and 3b, but for another embodiment which not only up-samples the samples of the input signal which are directly next to the interpolation moment I1, but also up-samples its neighboring signal samples. This is necessary when the signal in their neighbors also contribute to further processing. To the left in FIG. 5a, input samples arrive at a sample rate, and at the parallel time axis T, the interpolation moment I1 is indicated. The input signal is then up-sampled by a factor of 2 in a first up-sampling process, where a first disable/enable decision DE_E_A is performed in response to the interpolation moment I1, such that, in the illustrated case, two input samples at both sides of the interpolation moment are involved in the up-sampling process. To the right of Fig. 5a, the resulting signal after the up-sampling process, including three interpolated samples indicated with 'x'. Next, left side of Fig. 5b illustrates, by means of the two double-arrows, the input samples around the interpolation moment I1 which are included in a second up-sampling by a factor of 2. As seen, three samples on each side of the interpolation moment I1 are decided by a second disable/enable decision DE_E_B to form part of the second up-sampling process. To the right of FIG. 5b, the resulting samples after the second up-sampling are seen, thus forming the basis for the final calculation of the value representing the input signal at the interpolation moment I1.

FIG. 6 illustrates the 13-tap FIR low-pass filter used in the embodiment of FIGs. 5a and 5b, and serving to provide a correct phase representation. It requires 3 neighboring sampled points at each side to calculate the value of each up-sampled point. To ensure sufficient points for the second stage up-sampling, the neighboring signals of the interpolation time should also be up-sampled at the first stage up-sampling. In the first stage, FIG. 5a, the interpolation time I1 is between samples s3 and s4, but the signal is up-sampled between s2 and s5, instead of only s3 and s4, because the low pass filter in the second stage needs u1 and u3 as well for the calculation. Since samples s1 and s6 are irrelevant to the later interpolation process, they can be skipped. In the second stage, samples u1, s3, u2, s4, u3 and s5 are used to calculate a second stage up-sampling point by convoluting with the filter in FIG. 6.

FIG. 7 shows another cascaded structure suitable for software up-sampling. An input sequence $i_1..i_6$ is first 4 times up-sampling with up-sampling filter USF_1, providing an up-sampled output sequence $i_{0_0}..i_{1_2}$. Only a sub sequence of this output sequence $i_{0_0}..i_{1_2}$ is then identified as relevant for further processing, namely the samples in the dashed boxes, since they temporally span respective two interpolation times. Thus, only the identified samples are applied to the second up-sampling stage, including up-sampling filter USF_2,. Finally, the second up-sampling is performed and results in the output sequence $i_{0_0_0}..i_{1_1_1}$. Thus, in this embodiment a local up-sampling is preformed in the second stage, if it is assumed that the input data rate and the interpolation data rate is the same. In the first stage, there is an interpolation point for almost any two adjacent input samples. Therefore, it is no gain for the local up-sampling at this stage. However, after the first up-sampling, the output data rate of the first stage is 4 times higher than interpolation data rate. In this case, up-sampling only a part of the output samples from the first stage can save most of computations at the second stage. The output sequence, corresponding to SQ2 of Fig. 2, is then finally applied to a not shown interpolation algorithm.

FIG. 8 shows a device embodiment in the form of a shape sensing device, preferably arranged for real-time functioning, e.g. as part of a medical device. An optical sensing fiber OSF, i.e. a longitudinally shaped optical fiber or bunch of fibers, is optically connected to a console with a processor serving to control an optical source and sensor OSS which transmits optical energy to the optical sensing fiber OSF and receives the response there from. A real-time signal $s(t)$ in response to the shape of the optical sensing fiber OSF is then digitally sampled and up-sampled U_A adaptively according to the method as explained above. In the illustrated embodiment, the processor is then capable of calculating real-time values with correct phase representation, thus allowing calculation of the physical shape of the optical sensing fiber OSF, and in this embodiment providing an output to a display DP indicative of the shape of the optical sensing fiber.

FIG. 9 shows the optical network part of an optical shape sensing system. A sweep laser SL scans through certain wavelength range, typically between 1530nm and 1550nm. The output light of the laser SL is distributed to multiple branches of the optical network through splitters (oval shapes) and circulators CL. In the upper part of the network, an auxiliary interferometer AUX is used to trace the scan speed (frequency change) of the laser SL. The lower part of the network comprises 4 interferometers associated to the optical sensing fiber OSF. Each of these interferometers is connected to two detectors through a polarization beam splitter PBS. The detected signals of the lower part (called data signals)

need to be linearized with respect to the laser SL scan. That is, any two samples should correspond to the same amount of frequency change of the laser SL. One way of linearization can be done as follows:

- 1) Sampling all (aux and data) signals at a constant speed.
- 2) The interpolation points of data signal are calculated from the aux signal.
- 3) All data signals are re-sampled at the interpolation points

In a specific embodiment, for each data signal, two cascaded 4 times up-sampling processes have been implemented in a Virtex 5 SX95 on a NI PXIe-7966R board. The low pass filters used for the up-sampling are the ones shown in FIGs. 4a and 4b. The sample rate of each signal is 50MS/s. The output data rate of the first stage filter is about 200MS/s (4x50MS/s). If non-adaptive up-sampling is applied to the second stage, the data that goes through the low pass filter of the second stage has a rate of about 800MS/s. Consequently, the total data rate of 8 data signals is 6.4GS/s. Since the length of the low pass filter is 33, the filter of FIG. 4b, it requires about 200G multiplications per second in total. Even with an optimized algorithm [Mou, Z.-J. (1996). *Symmetry exploitation in digital interpolators/decimators*. *IEEE Transactions on Signal Processing*, 44 (10), 2611 - 2615], the number of multiplications for the second stage filters is still at the rate of 76.8G/s, which are not able to be handled by the provided FPGA. Since the interpolation times derived from the aux signal has a rate of about 50MS/s, only 25% of the input data needs to go through the low pass filter in case of the adaptive up-sampling. Therefore, the total number of multiplications for the second stage filters is about 19.2G/s, which can be successfully implemented on the provided FPGA.

FIG. 10 illustrates steps of a method embodiment. First, an input sequence of samples at a sample rate $Q1$ is received R_IS_Q1 , and a signal indicative of interpolation times is also received R_IM_QM , at a sample rate QM which is preferably smaller than $Q1$. Next, a limited number of the samples in the input sequence are identified I_LS adaptively to the received interpolation times, such that only temporally relevant samples are selected, e.g. neighboring samples around the interpolation times. Next, the limited number of samples is then up-sampled US_LS_Q2 to a sample rate $Q2$ which is preferably an integer, e.g. 2, 3, 4 or more, of the sample rate $Q1$ of the input sequence. To preserve correct representation of phase in the resulting up-sampled output signal, the up-sample process US_IM_Q2 preferably involves low pass filtering the input samples with a linear FIR filter, e.g. in the form of a Sinc function. Next, the method comprises interpolating samples in the up-sampled signal around the interpolation times INT , e.g. by linear interpolation. Finally, the method

comprises calculating a value representative of the input signal to a time corresponding to the interpolation times C_V_IM . Preferably, the method is arranged for being performed in real-time, i.e. outputting a stream of output values corresponding to interpolation moments.

It is to be understood that the method steps do not necessarily have to be carried out in the illustrated order.

To sum up, the invention provides a method and device for calculating a value representative of a signal at an interpolation time $SQTM$, based on an input sequence $SQ1$ representative of a temporal sampling of the signal. An adaptive up-sampling is performed by identifying a limited portion of the input sequence SS , selected such as to temporally span the interpolation time $SQTM$, thus allowing precise interpolation of input samples temporally local to the interpolation time $SQTM$. The limited portion of the input sequence SS , e.g. only one sample on each side of the interpolation time $SQTM$, is then up-sampled to form an up-sampled sequence $SQ2$, which has a rather low rate on average, due to the discarded input samples, e.g. equal to the sample rate of the input sequence $SQ1$. However, locally around the interpolation time $SQTM$, the sample rate is higher. Finally, it is possible to calculate a value representative of the signal at the interpolation time $V(TM)$ in response to the up-sampled sequence $SQ2$. Using linear FIR filters in the up-sampling process, a significant reduction of processing power can be saved, and still a high precision with a preserved phase information can be obtained, thus allowing the method to be used e.g. for optical shape sensing applications.

While the invention has been illustrated and described in detail in the drawings and foregoing description, such illustration and description are to be considered illustrative or exemplary and not restrictive; the invention is not limited to the disclosed embodiments. Other variations to the disclosed embodiments can be understood and effected by those skilled in the art in practicing the claimed invention, from a study of the drawings, the disclosure, and the appended claims. In the claims, the word "comprising" does not exclude other elements or steps, and the indefinite article "a" or "an" does not exclude a plurality. A single processor or other unit may fulfill the functions of several items recited in the claims. The mere fact that certain measures are recited in mutually different dependent claims does not indicate that a combination of these measures cannot be used to advantage. A computer program may be stored/distributed on a suitable medium, such as an optical storage medium or a solid-state medium supplied together with or as part of other hardware, but may also be distributed in other forms, such as via the Internet or other wired or wireless

telecommunication systems. Any reference signs in the claims should not be construed as limiting the scope.

CLAIMS:

1. A method for calculating a value representative of a signal at an interpolation time (SQT_M), based on an input sequence (SQ₁) representative of the signal, the method comprising
 - receiving the input sequence (SQ₁), wherein the input sequence (SQ₁) comprises a series of input samples (s₁, s₂, s₃) temporally sampled,
 - receiving information about an interpolation time (SQT_M),
 - identifying a limited portion of the input sequence (SS), wherein the limited portion of the input sequence (SS) is selected in response to the interpolation time (SQT_M) so as to temporally span the interpolation time (SQT_M),
 - up-sampling the limited portion of the input sequence (SS) to an up-sampled sequence (SQ₂), and
 - calculating a value representative of the signal at the interpolation time (V(T_M)) in response to the up-sampled sequence (SQ₂).
2. Method according to claim 1, wherein the limited portion of the input sequence (SS) contains only input samples temporally near to the interpolation time (SQT_M).
3. Method according to claim 1, wherein the limited portion of the input sequence (SS) contains only input samples temporally adjacent to the interpolation time (SQT_M).
4. Method according to claim 1, wherein the limited portion of the input sequence (SS) contains only a first plurality of input samples at times lower than the interpolation time (SQT_M) and a second plurality of input samples at times higher than the interpolation time (SQT_M).
5. Method according to claim 1, wherein a sample rate of the up-sampled sequence (SQ₂) is an integer times a sample rate of the input sequence (SQ₁) locally around the interpolation time.

6. Method according to claim 1, comprising identifying a limited portion of the up-sampled sequence, wherein the limited portion of the up-sampled sequence (SS) is selected in response to the interpolation time so as to temporally span the interpolation time, and up-sampling the limited portion of the up-sampled sequence to a second up-sampled sequence, and calculating a value representative of the signal at the interpolation time in response to the second up-sampled sequence.
7. Method according to claim 1, wherein the up-sampling comprises convoluting with a Finite Impulse Response low pass filter.
8. Method according to claim 7, wherein the Finite Impulse Response filter is a linear Finite Impulse Response low pass filter.
9. Method according to claim 1, comprising receiving the information about interpolation time (SQTM) at an interpolation rate.
10. Method according to claim 9, wherein the interpolation rate is lower than a sample rate of the input sequence (SQ1).
11. Method according to claim 10, wherein the interpolation rate is 1-25% of the sample rate of the input sequence (SQ1).
12. Method according to claim 1, comprising receiving information about a plurality of interpolation times, and identifying a limited portion of the input sequence, wherein the limited portion of the input sequence is selected in response to the plurality of interpolation times so as to form groups of the input samples temporally spanning the respective plurality of interpolation times, up-sampling the limited portion of the input sequence to an up-sampled sequence, and calculating values representative of the signal at the plurality of interpolation times in response to the up-sampled sequence.
13. Method according to claim 1, wherein calculating the value representative of the signal at the interpolation time comprises applying an interpolation algorithm to the up-sampled sequence.

14. A device comprising a processor (P) arranged
- to receive an input sequence comprising a series of input samples temporally sampled,
 - to receive information about an interpolation time,
 - to identify a limited portion of the input sequence, wherein the limited portion of the input sequence is selected in response to the interpolation time so as to temporally span the interpolation time,
 - to up-sample the limited portion of the input sequence to an up-sampled sequence, and
 - to calculate a value representative of the signal at the interpolation time in response to the up-sampled sequence.
15. Device according to claim 14, wherein the device comprises an optical shape sensor (OSF), wherein the signal ($s(t)$) is indicative of a shape of an optical shape sensor (OSF), and wherein the device is arranged to update a measure of shape of the optical shape sensor (OSF) in response to the calculated value representative of the signal ($s(t)$) at the interpolation time.

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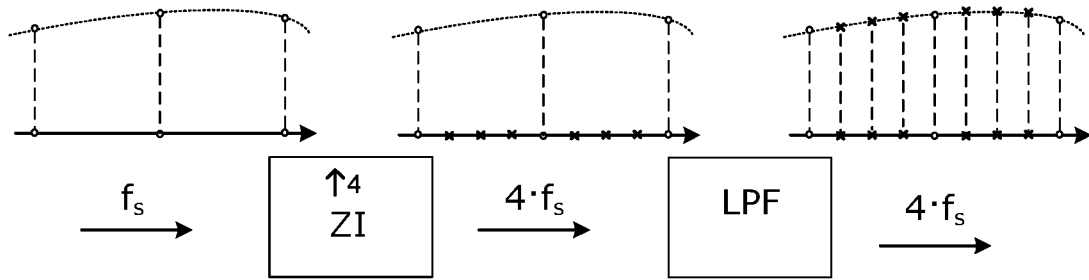
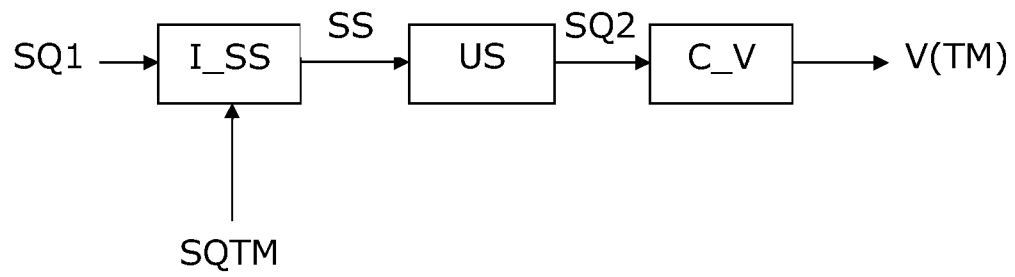
Prior art
FIG. 1

FIG. 2

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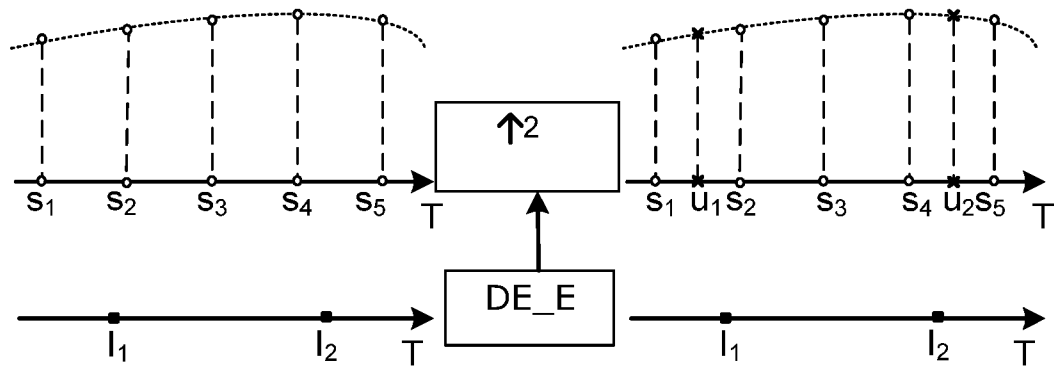


FIG. 3a

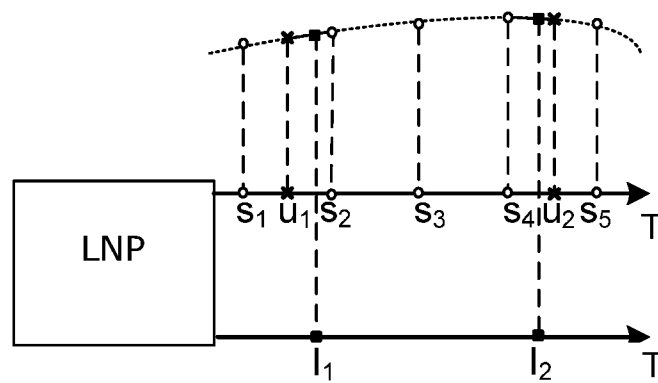


FIG. 3b

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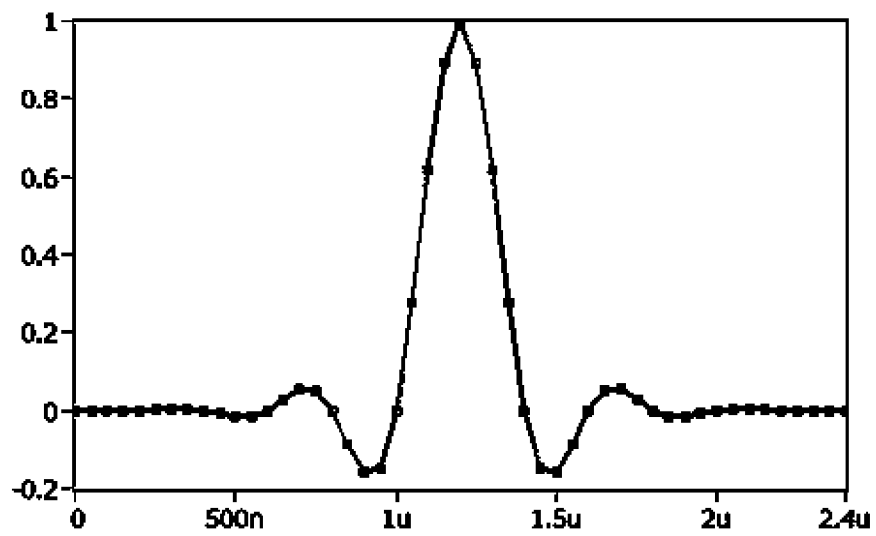


FIG. 4a

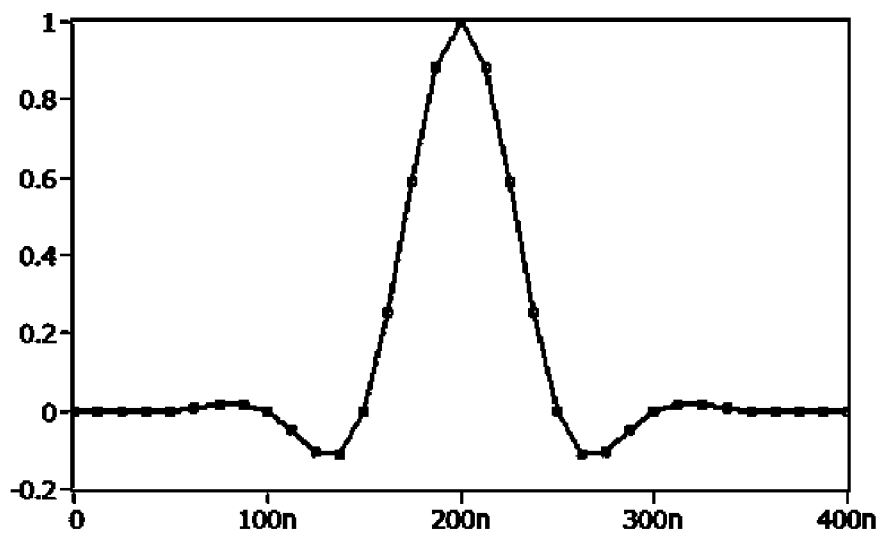


FIG. 4b

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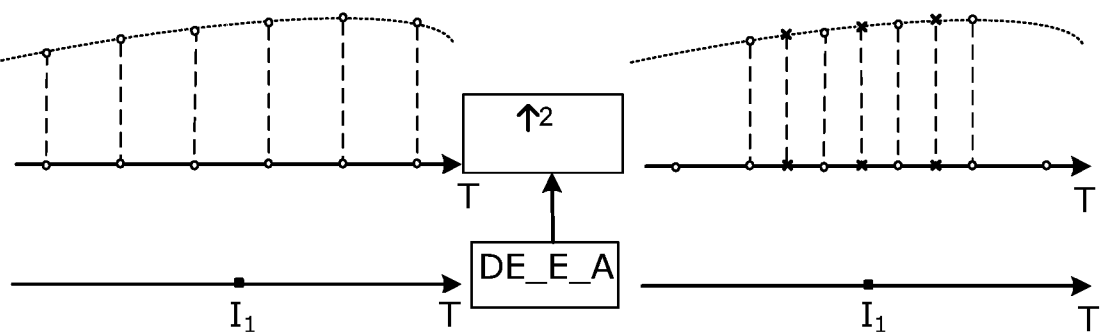


FIG. 5a

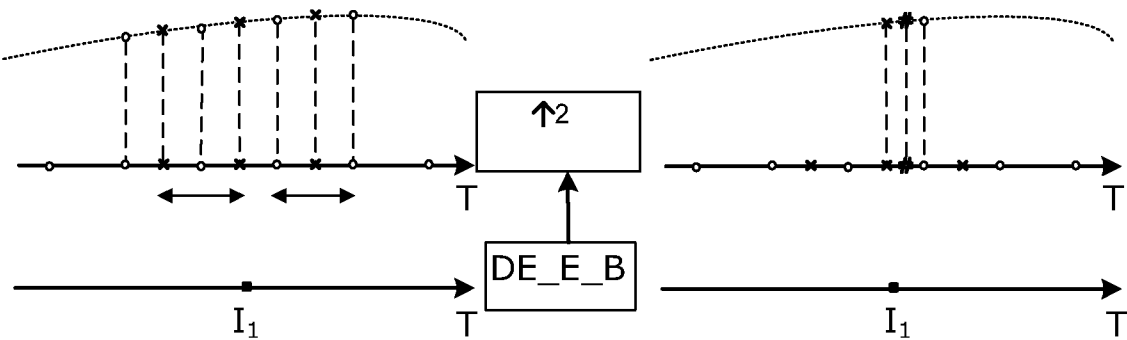


FIG. 5b

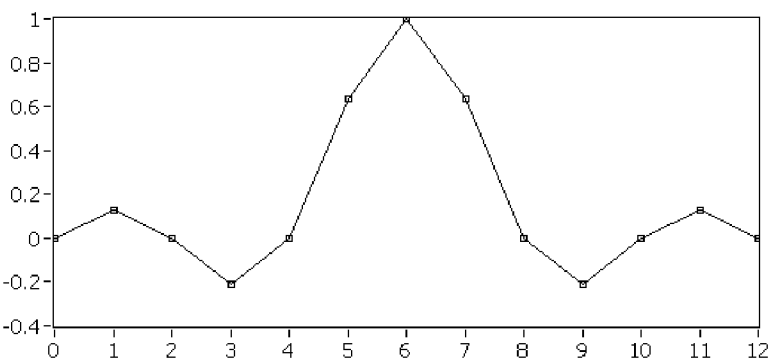


FIG. 6

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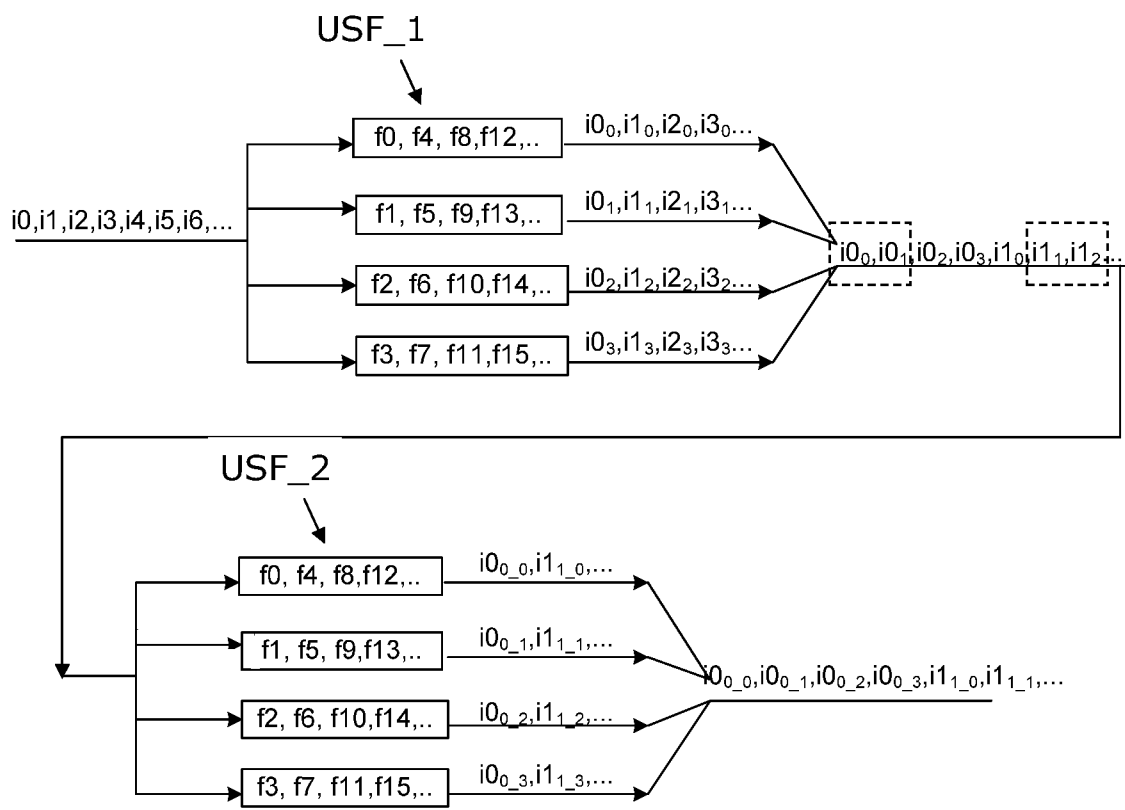


FIG. 7

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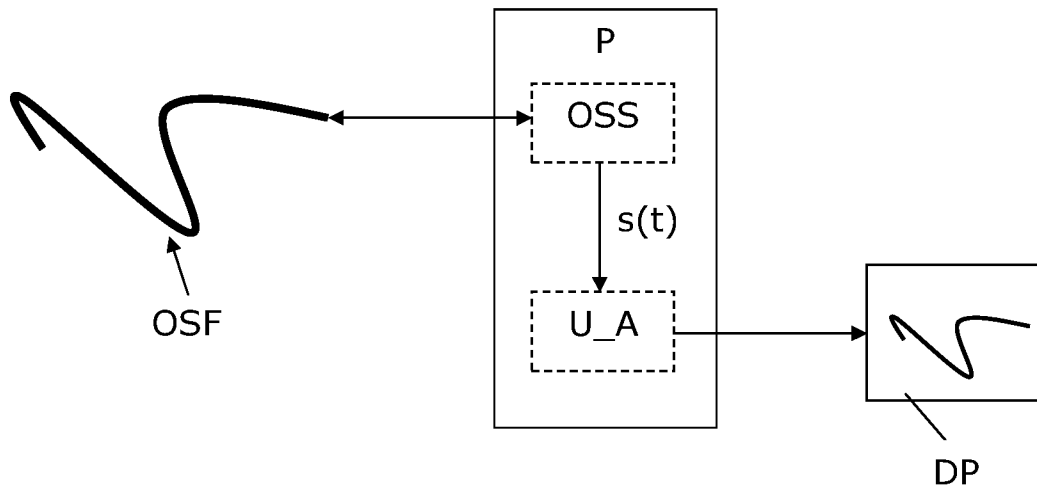


FIG. 8

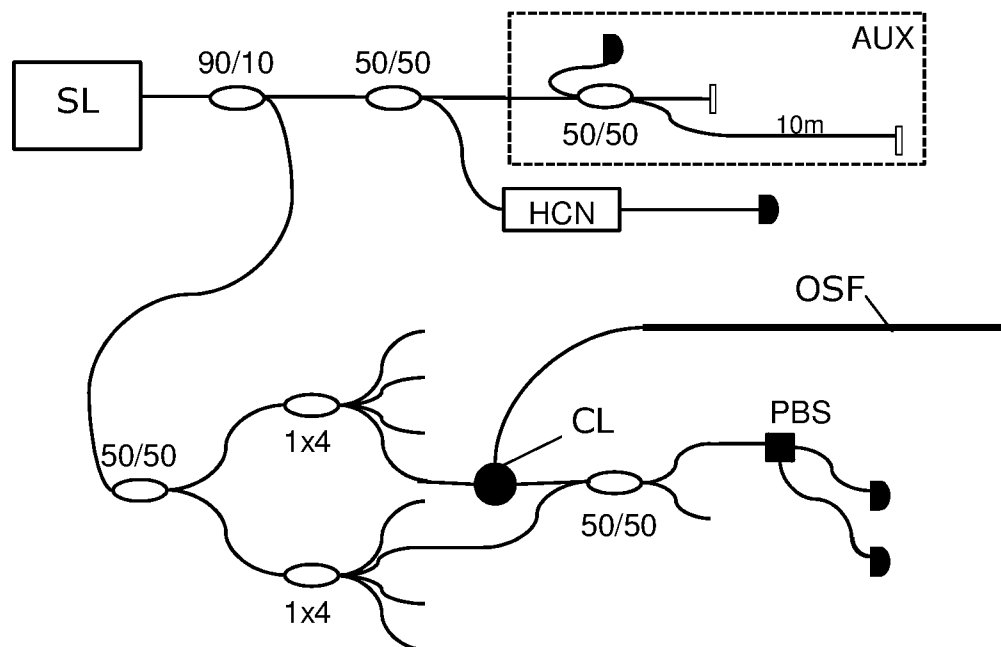


FIG. 9

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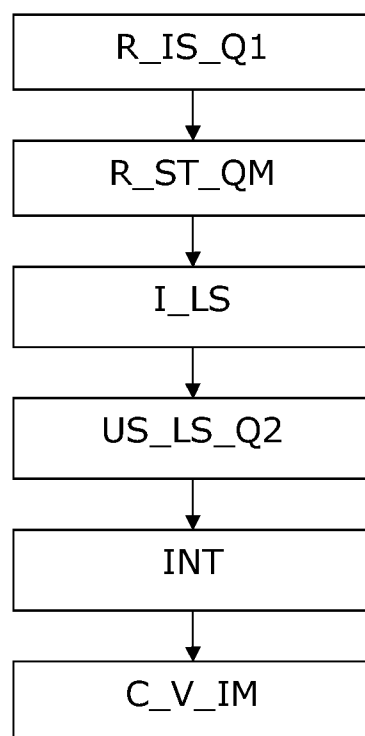


FIG. 10