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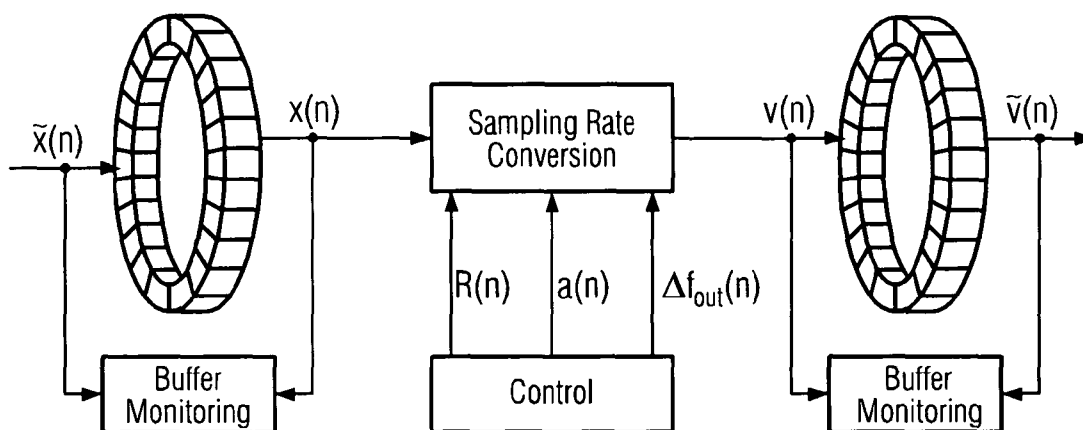
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(54) **Sampling rate conversion**

(57) The present invention relates to a method for processing an input signal  $x(n)$  of a first sampling rate that comprises the steps of up-sampling (1) the input signal  $x(n)$  to obtain an up-sampled signal  $x_L(n)$ , low-pass filtering (2) the up-sampled signal  $x_L(n)$  to obtain a first filtered up-sampled signal  $y_L(n)$ , time-delaying (3) the up-sampled signal  $x_L(n)$  and subsequently low-pass filtering (4) the time-delayed up-sampled signal  $x_L$

( $n-1$ ) to obtain a second filtered up-sampled signal  $y_L(n-1)$ , weighting the first filtered up-sampled signal  $y_L(n)$  by a first weight factor to obtain a first weighted signal, weighting the second filtered up-sampled signal  $y_L(n-1)$  by a second weight factor different from the first weight factor to obtain a second weighted signal, and adding the first weighted signal and the second weighted signal to obtain an output signal  $v_L(n)$  of a second sampling rate different from the first sampling rate.



**FIG. 3**

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**Description**

**Field of Invention**

5 [0001] The present invention relates to the processing of band-limited signals and is particularly concerned with the conversion of electrical signals sampled with a first sampling rate to electrical signals with a second sampling rate that is different from the first sampling rate. More particular, the invention relates to the sampling rate conversion in the case of a temporarily varying destination sampling rate.

10 **Prior Art**

[0002] The conversion of sampling rates is a major problem in signal processing. When two electrical devices communicate by the exchange of electrical signals usually the problem arises that the sampling rate of the one device is different from the sampling rate of the other one. For example, a first device may detect signals with a first sampling rate and process these signals. After processing, these signals are output to a second device. The second device may be intended to process the input signals coming from the first device and to output them with a sampling rate higher than the sampling rate of the first device. In this case some up-sampling must be carried out by the second device. One example of such a communication is the exchange of signals between a cellular phone and a hands-free set via the Bluetooth technology.

20 [0003] Another example for the need for some up-sampling of electrical signals is given by the extension of band-limited signals in the context of telephony. The common telephone band-limited audio and, in particular, speech signals exhibit a bandwidth of only 300 Hz to 3.4 kHz. Since the removal of portions of live signals with lower and higher frequencies causes degradation in speech quality, in particular, a reduced intelligibility, it is highly desirable to extend the limited bandwidth to a wider bandwidth. Some preprocessing, as increasing the sampling rate by interpolation, must usually be performed before analyzing the band limited signal in order to generate an extended less artificial signal.

25 [0004] The increase of a given sampling rate, i.e. the number of individual samples taken per unit time from a continuous signal to generate a discrete signal, requires the interpolation of the signal between the discrete time samples. Well-known interpolation methods include Lagrange interpolation, which is the classical technique of finding an order n polynomial which passes through n+1 given data points. Another technique known as Cubic Splines fits a third order polynomial through two points so as to achieve a certain slope at one of the data points.

30 [0005] In particular, in the context of sampling rate conversion with a temporarily varying destination sampling rate, the following two methods have been employed in the past. According to the first method, a highly over-sampled signal is generated from an input signal. For each destination discrete sample, the closest artificially generated time sample is chosen. The half of the over-sampling rate represents the maximum error. Obviously, the higher the over-sampling rate is chosen the lower the corresponding error is. This brute force method, however, requires a high computing and memory capacitance.

35 [0006] The second method represents a quasi-continuous interpolation of individual samples  $x(n)$  of a continuous absolutely integrable signal  $x(t)$ , where  $t$  is time in seconds and  $n$  ranges over integers. According to this method, the sine cardinal function is employed in order to generate a signal value at the desired sampling time  $T_a$ :

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$$x(T_a) = \sum_{n=-\infty}^{\infty} x(n)h_s(T_a - nT_{in}), \tag{1}$$

45 with

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$$h_s(t) = \text{sinc}(f_{in}t) := \frac{\sin(\pi f_{in}t)}{\pi f_{in}t}. \tag{2}$$

55 [0007] In these equations  $T_{in}$  and  $f_{in}$  represent the original (input) sampling time (period) and sampling rate ( $f_{in} = 1 / T_{in}$ ), respectively.

[0008] If the destination sampling rate is less than the original sampling rate, a low pass cut-off must be placed below half of the new lower sampling rate in order to avoid aliasing.

[0009] Whereas the sinc-interpolation guarantees a relatively high quality for an arbitrary choice of the destination sampling rate, it also requires a huge amount of computing time. For a detailed interpolation of the sinc-function, high memory resources are required. Moreover, in practice the quality of the reproduction signal is limited by the necessity of a finite summation in equation 1.

[0010] It is therefore, the problem underlying the present invention to overcome the above-mentioned drawbacks and to provide a method for the conversion of sampling rates with a smaller requirement for computer resources as compared to the art.

**Description of the Invention**

[0011] The above-mentioned problem is solved by a method according to claim 1. The inventive method for processing an input signal of a first sampling rate comprises the steps of up-sampling the input signal to obtain an up-sampled signal; low-pass filtering the up-sampled signal to obtain a first filtered up-sampled signal; time-delaying the up-sampled signal and subsequently low-pass filtering the time-delayed up-sampled signal to obtain a second filtered up-sampled signal; weighting the first filtered up-sampled signal by a first weight factor to obtain a first weighted signal; weighting the second filtered up-sampled signal by a second weight factor different from the first weight factor to obtain a second weighted signal; and adding the first weighted signal and the second weighted signal to obtain a first output signal of a second sampling rate different from the first sampling rate.

[0012] The input signal of a first sampling rate may be a digital signal sampled from an analogous signal with a predetermined sampling rate, i.e. number of individual samples per time taken from a continuous signal to generate the discrete digital signal. The input signal may comprise more or less sampling instants and, thus, individual samples, per band interval than the first output signal.

[0013] The inventive method only requires for relatively small computer resources, in particular, memory. Therefore, the method is very practical for hardware implementation. Basically, it comprises just one convolution of low order and a few addition operations.

[0014] Low-pass filtering of both the up-sampled and the time-delayed up-sampled signals are preferably carried out by means of a Finite Impulse Response (FIR) filter in order to delete aliasing components that otherwise might occur.

[0015] The low-pass filtering of the time-delayed up-sampled signal can be performed using the same impulse response as for the low-pass filtering of the up-sampled signal, but delayed according to the time-delay of the signal. Advantageously, the time delaying step results in a time delay of one sampling instant (one discrete sample). In this case, low-pass filtering of the time-delayed up-sampled signal is carried out by the impulse response used for filtering the non-delayed signal but delayed by one sampling instant.

[0016] The first and second weight factors may be time-dependent. The value of the second weight factor may be given by 1 - the value of the first weight factor. Thus, with  $y_L(n)$  and  $y_L(n-1)$  denoting the first and second filtered up-sampled signal, respectively, and  $a(n)$  denoting the first weight factor the first output signal  $v_L(n)$  is obtained by  $v_L(n) = a(n) y_L(n) + (1 - a(n)) y_L(n-1)$ .

[0017] The low-pass filtering can very effectively be carried out when the up-sampled signal is obtained by inserting  $L - 1$  nulls between the individual samples for the respective sampling instants of the input signal of the first sampling rate (i.e., inserting nulls, if  $\text{mod}(n, L) \neq 0$ , where  $n$  ranges over integers and represents a discrete time index, i.e. sampling instant), where  $L$  is the factor by which the first signal is up-sampled, and a modified convolution of the input signal with the impulse response of the low-pass FIR filter  $h_{TP,i}$  according to

$$y_L(n) = \sum_{k=0}^{\lfloor \frac{N}{L} \rfloor} x(\lfloor n/L \rfloor - k) h_{TP, kL + \text{mod}(n, L)} \cdot \tag{3}$$

[0018] In equation 3 the parameter  $N$  denotes the length (number of filter coefficients) of the low-pass filter.

[0019] In this case, the first output signal of the second sampling rate can be obtained by means of time-dependent filter coefficients  $h_{TP}(n)$  used for the low-pass filtering according to

$$v_L(n) = a(n)y_L(n) + (1 - a(n))y_L(n - 1) = \mathbf{x}^T(n)\mathbf{h}_{TP}(n) \quad , \quad (4)$$

5

where the upper index T indicates the transposition operation, and with

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$$\mathbf{x}(n) = \begin{bmatrix} x(\lfloor n/L \rfloor) \\ x(\lfloor n/L \rfloor - 1) \\ \dots \\ x(\lfloor n/L \rfloor - \lceil N/L \rceil) \end{bmatrix} \quad (5)$$

15

and

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$$\mathbf{h}_{TP}(n) = a(n) \begin{bmatrix} h_{TP, \text{mod}(n, L)} \\ h_{TP, L + \text{mod}(n, L)} \\ \dots \\ h_{TP, L \lceil N/L \rceil + \text{mod}(n, L)} \end{bmatrix} + (1 - a(n)) \begin{bmatrix} h_{TP, \text{mod}(n, L) - 1} \\ h_{TP, L + \text{mod}(n, L) - 1} \\ \dots \\ h_{TP, L \lceil N/L \rceil + \text{mod}(n, L) - 1} \end{bmatrix} \quad (6)$$

30

[0020] Usually, if some up-sampling of an input signal is desired, the first output signal after up-sampling exhibits a sampling rate above the eventually desired one. Therefore, the first output signal ( $v_L(n)$ ) of the second sampling rate can be down-sampled to obtain a second (destination) output signal ( $v(n)$ ) of a third sampling rate that is different from the second sampling rate and usually also different from the first sampling rate.

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[0021] In this case, the first weight factor can be obtained by

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$$a(n) = 1 - \left( \left\lceil n \frac{f_{in}}{f_{out}} L \right\rceil - n \frac{f_{in}}{f_{out}} L + f_0 \delta_K(n) \right) \quad (7)$$

45

with  $f_0$  defined as

50

$$f_0 = \left( \left\lceil \frac{f_{in}}{f_{out}} L \right\rceil - \frac{f_{in}}{f_{out}} L \right), \quad (8)$$

55

where  $\delta_K$  is the Kronecker-Delta function and  $L$  is the factor by which the input (original) signal of the first (original) sampling rate  $f_{in}$  is up-sampled to the third sampling rate  $f_{out}$  of the second (destination) output signal. The symbols  $\lceil \cdot \rceil$  and  $\lfloor \cdot \rfloor$  denote rounding to the closest larger or smaller integer, respectively.

[0022] The above-described aspects of the inventive method can also be modified in order to be useful for a time-dependent third sampling rate to which the input signal is to be converted. When the third sampling rate is time-dependent, the input signal of the first sampling rate can be stored in a first buffer, in particular, a first ring buffer, and/or the second

output signal of the third sampling rate can be stored in a second buffer, in particular, a second ring buffer. By monitoring the respective write and read pointers it can be determined, whether one of the buffers becomes empty or is close to an overflow state. In these cases, the third sampling rate can be corrected in order to guarantee faultless processing of the input signal.

[0023] The time-dependent third sampling rate may comprise a constant nominal value as well as a time-dependent correction term  $\Delta f_{out}(n)$  that can be adjusted according to

$$\Delta f_{out}(n) = \begin{cases} \min\{\Delta f_{out}(n-1) + \Delta, \Delta f_{out,max}\}, & \text{if content of second buffer} \\ & \text{is below a first predetermined} \\ & \text{threshold} \\ \max\{\Delta f_{out}(n-1) - \Delta, \Delta f_{out,min}\}, & \text{if content of second buffer} \\ & \text{is above a second predetermined} \\ & \text{threshold} \\ \Delta f_{out}(n-1), & \text{else} \end{cases}$$

(9)

where  $\Delta$  is the maximum change of the sampling rate from one sampling instant to the subsequent sampling instant.

[0024] Employing the correction term  $\Delta f_{out}(n)$  given in equation 9, the second output signal  $v(n)$  of the third sampling rate can be obtained by

$$v(n) = \left( 1 - \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] + \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L - f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) \right) + \left( \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] - \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L + f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) - 1 \right)$$

(10)

with the down-sampling factor

$$R(n) = \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] - \sum_{m=0}^{n-1} R(m) + \left[ \frac{f_{in}}{f_{out}} L \right] \delta_K(n).$$

(11)

[0025] Embodiments of the inventive methods that are useful for sampling rate conversion in the case of a time-dependent third (destination) sampling rate can also be employed in cases in which the first (original) sampling rate is time-dependent.

[0026] The present invention also provides a signal processing means configured to generate from an input signal of a first sampling rate a first output signal of a second sampling rate different from the first sampling rate and/or configured to generate from an input signal of a first sampling rate a second output signal of a third sampling rate different from the first sampling rate and/or different from the second sampling rate by carrying out processing steps according to one of the above-discussed examples of the inventive method.

[0027] Moreover, an electrical device is provided that is configured to send signals to and/or receive signals from

another electrical device, comprising the signal processing means mentioned above. Examples for the electrical device are cellular phones, headsets, and hands-free sets.

[0028] Furthermore, the present invention provides a computer program product comprising one or more computer readable media having computer executable instructions for performing the steps of embodiments of the inventive method for sampling rate conversion as described above.

[0029] Additional features and advantages of the present invention will be described with reference to the drawings. In the following description, reference is made to the accompanying figures that are meant to illustrate preferred embodiments of the invention. It is understood that such embodiments do not represent the full scope of the invention that is defined by the claims given below.

[0030] Fig. 1 is a block diagram illustrating the principles of an example of the method for sampling rate conversion according to the present invention including low-pass filtering of an up-sampled original signal and a time-delayed up-sampled original signal and weighting of the low-pass filtered signals.

[0031] Fig. 2 illustrates the original signal, time-dependent weighting, time-dependent sampling rate decimation and the destination signal.

[0032] Fig. 3 illustrates an example for the sampling rate conversion according to the present invention for the case of a time-dependent destination sampling rate. This example employs buffering of the original as well as the destination signal.

[0033] Fig. 1 illustrates an example of the inventive method of sampling rate conversion to obtain an output signal  $v(n)$ , where  $n$  denotes the discrete time index, of a destination (output) sampling rate  $f_{out}$  from an input signal  $x(n)$  of an original (input) sampling rate  $f_{in}$ . First, the input signal  $x(n)$  is up-sampled 1 to a sampling rate  $f_{up} > f_{in}$  by a factor of  $L$ ,  $L$  being an integer, i.e.  $f_{up} = L f_{in}$ . The up-sampled signal  $x_L(n)$  is obtained by inserting  $L - 1$  nulls between the samples of the original signal, i.e.  $x_L(n) = x(n/L)$ , if  $\text{mod}(n, L) = 0$ ,  $\text{mod}$  being the modulo function, and  $x_L(n) = 0$ , else.

[0034] On the one hand,  $x_L(n)$  is subject to a convolution 2 with an impulse response of a low-pass filter with coefficients  $h_{TP,i}$  in one processing branch

$$y_L(n) = \sum_{i=-\infty}^{\infty} x_L(n-i)h_{TP,i} .$$

[0035] On the other hand,  $x_L(n)$  is time-delayed 3 by one sampling instant in another processing branch and the time-delayed signal is subsequently subject to a convolution 4 with the impulse response comprising the same filter vector as employed in the first processing branch but delayed by one sampling instant. In order to prevent aliasing, an FIR filter is used for the low-pass filtering process.

[0036] Further, since  $x_L(n)$  has been generated by inserting nulls, the filtered non-delayed up-sampled signal  $y_L(n)$  can be obtained by

$$y_L(n) = \sum_{k=0}^{\left\lfloor \frac{N}{L} \right\rfloor} x_L(n - kL - \text{mod}(n, L))h_{TP, kL + \text{mod}(n, L)}$$

and since  $x_L(n)$  is only used for non-vanishing values

$$y_L(n) = \sum_{k=0}^{\left\lfloor \frac{N}{L} \right\rfloor} x(\lfloor n/L \rfloor - k)h_{TP, kL + \text{mod}(n, L)} ,$$

where the symbols  $\lceil \rceil$  and  $\lfloor \rfloor$  denote rounding to the closest larger or smaller integer, respectively.

[0037] A control means 5 provides time-dependent factors  $a(n)$  and  $1 - a(n)$  for the first and second processing branch, respectively, to obtain the output signal  $v_L(n)$  of a sampling rate much higher than the one of the input signal by the weighted sum of  $y_L(n)$  and  $y_L(n-1)$ :

$$v_L(n) = a(n)y_L(n) + (1 - a(n))y_L(n-1).$$

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[0038] The weighting and convolution processes can be carried out by

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$$v_L(n) = \mathbf{x}^T(n) \mathbf{h}_{TP}(n)$$

by means of the signal vector

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$$\mathbf{x}(n) = \begin{bmatrix} x(\lfloor n/L \rfloor) \\ x(\lfloor n/L \rfloor - 1) \\ \dots \\ x(\lfloor n/L \rfloor - \lceil N/L \rceil) \end{bmatrix}$$

20

25 and the time-dependent filter vector of the used low-pass filter

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$$\mathbf{h}_{TP}(n) = a(n) \begin{bmatrix} h_{TP, \text{mod}(n,L)} \\ h_{TP, L + \text{mod}(n,L)} \\ \dots \\ h_{TP, L \lceil N/L \rceil + \text{mod}(n,L)} \end{bmatrix} + (1 - a(n)) \begin{bmatrix} h_{TP, \text{mod}(n,L)-1} \\ h_{TP, L + \text{mod}(n,L)-1} \\ \dots \\ h_{TP, L \lceil N/L \rceil + \text{mod}(n,L)-1} \end{bmatrix}.$$

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[0039] By an appropriate choice of a(n) by the processing means 5, in principle, the signal value for each arbitrary sampling instant, i.e. the discrete time point at which an individual sample of a signal is taken, can be calculated at least approximately.

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[0040] The sampling rate  $f_{up}$  of the output signal  $v_L(n)$  is usually higher than a desired sampling rate  $f_{out}$  and, thus,  $v_L(n)$  must be down-sampled. In particular, by an appropriate choice of a(n) and the parameters of a down-sampling factor  $R(n)$  it can be ensured that each sampling instant of the original signal exactly corresponds to one sampling instant of a desired sampling rate as follows:

45

$$R(n) = \left\lfloor n \frac{f_{in}}{f_{out}} L \right\rfloor - \sum_{m=0}^{n-1} R(m) + \left\lfloor n \frac{f_{in}}{f_{out}} L \right\rfloor \delta_K(n)$$

50

and

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$$a(n) = 1 - \left( \left\lfloor n \frac{f_{in}}{f_{out}} L \right\rfloor - n \frac{f_{in}}{f_{out}} L + f_0 \delta_K(n) \right)$$

with the Kronecker-Delta function  $\delta_K$  and where  $f_0$  is defined as

$$f_0 = \left( \left[ \frac{f_{in}}{f_{out}} L \right] - \frac{f_{in}}{f_{out}} L \right).$$

[0041] Thus, the desired output signal  $v(n)$  with a sampling rate higher than the input signal  $x(n)$  but lower than the sampling rate of  $v_L(n)$  is eventually obtained as

$$v(n) = \left( 1 - \left[ n \frac{f_{in}}{f_{out}} L \right] + n \frac{f_{in}}{f_{out}} L - f_0 \delta_K(n) \right) y_L \sum_{m=0}^n R(m) + \left( \left[ n \frac{f_{in}}{f_{out}} L \right] - n \frac{f_{in}}{f_{out}} L + f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) - 1 \right).$$

[0042] Fig. 2 shows an example of a sampling rate conversion of a signal with an original sampling rate of 8 kHz as indicated by the integer  $n$  (upper row) to a destination signal of a sampling rate of 11,025 kHz as indicated by the integer  $m$  (lower row). Eight sampling instants ( $n = 0 \dots 7$ ) are shown for the original signal  $x(n)$  corresponding to ten sampling instants ( $m = 0 \dots 9$ ) in a comparable band interval. The original signal is converted by means of the weighting factors  $a(m)$  and  $1 - a(m)$  (second row) and the time-dependent down-sampling  $R(m)$ . The factor for the up-sampling is set to  $L = 1$  for an easier understanding and for a better intelligibility of the second row the weighting factors are slightly shifted with respect to each other.

[0043] Fig. 3 shows another example for the inventive method of sampling rate conversion wherein the destination sampling rate is time-dependent. As shown in Fig. 3 the original (input) signal  $\tilde{x}(n)$  and the destination (output) signal  $v(n)$  are buffered in a ring buffer. Alternatively, only one of the original signal and the destination signal is input in a buffer. By monitoring the write/read pointers it can be determined whether one of the buffers becomes empty or runs the risk of an overflow. Correction of the destination sampling rate in order to avoid an empty or overflow state of one of the buffers can be performed as follows. The destination rate  $f_{out}$  is split in a constant part  $\bar{f}_{out}$  and a time-varying correction  $\Delta f_{out}(n) : f_{out} = \bar{f}_{out} + \Delta f_{out}(n)$  where

$$\Delta f_{out}(n) = \begin{cases} \min \{ \Delta f_{out}(n-1) + \Delta, \Delta f_{out,max} \}, & \text{if content of second buffer} \\ & \text{is below a first predetermined} \\ & \text{threshold} \\ \max \{ \Delta f_{out}(n-1) - \Delta, \Delta f_{out,min} \}, & \text{if content of second buffer} \\ & \text{is above a second predetermined} \\ & \text{threshold} \\ \Delta f_{out}(n-1), & \text{else} \end{cases}$$

and  $\Delta$  is the maximum change of the sampling rate from one sampling instant to the subsequent sampling instant and may, e.g., be in the range of  $10^{-7}$  to 0.01 Hz.

[0044] Using the time-dependent destination sampling rate including the correction term  $\Delta f_{out}(n)$  the calculations



described with reference to Fig. 1 are performed to obtain the

$$\begin{aligned}
 \mathbf{v}(n) = & \left( 1 - \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}} L \right] + \sum_{i=0}^n \frac{f_{in}}{f_{out}} L - f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) \right) \\
 & + \left( \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}} L \right] - \sum_{i=0}^n \frac{f_{in}}{f_{out}} L + f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) - 1 \right)
 \end{aligned}$$

with the down-sampling factor

$$R(n) = \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}} L \right] - \sum_{m=0}^{n-1} R(m) + \left[ \frac{f_{in}}{f_{out}} L \right] \delta_K(n).$$

**[0045]** In addition, the above-described signal processing may also be performed for a time-varying input sampling rate with which the digital input signal  $x(n)$  might be sampled from an continuous analogous signal.

## Claims

1. Method for processing an input signal ( $x(n)$ ) of a first sampling rate, comprising the steps of

up-sampling (1) the input signal ( $x(n)$ ) to obtain an up-sampled signal ( $x_L(n)$ );  
 low-pass filtering (2) the up-sampled signal ( $x_L(n)$ ) to obtain a first filtered up-sampled signal ( $y_L(n)$ );  
 time-delaying (3) the up-sampled signal ( $x_L(n)$ ), in particular, by one sampling instant, and subsequently low-pass filtering (4) the time-delayed up-sampled signal ( $x_L(n-1)$ ) to obtain a second filtered up-sampled signal ( $y_L(n-1)$ );  
 weighting the first filtered up-sampled signal ( $y_L(n)$ ) by a first weight factor to obtain a first weighted signal;  
 weighting the second filtered up-sampled signal ( $y_L(n-1)$ ) by a second weight factor different from the first weight factor to obtain a second weighted signal; and  
 adding the first weighted signal and the second weighted signal to obtain a first output signal ( $v_L(n)$ ) of a second sampling rate different from the first sampling rate.

2. The method according to claim 1, wherein the low-pass filtering (2, 4) of both the up-sampled and the time-delayed up-sampled signals ( $x_L(n)$ ,  $x_L(n-1)$ ) is carried out by means of a Finite Impulse Response Filter.

3. The method according to claim 1 or 2, wherein the time-delayed up-sampled signal ( $x_L(n-1)$ ) is time-delayed by one sampling instant with respect to the non-delayed up-sampled signal ( $x_L(n)$ ) and the low-pass filtering (4) of the time-delayed up-sampled signal ( $x_L(n-1)$ ) is performed using the same impulse response but delayed by one sampling instant as the one used for the low-pass filtering (2) of the non-delayed up-sampled signal ( $x_L(n)$ ).

4. The method according to one of the preceding claims, wherein the first and the second weight factors are time-dependent or constant and the value of the second weight factor is given by 1 minus the value of the first weight factor.

5. The method according to one of the claims 2 - 4, wherein the up-sampled signal ( $x_L(n)$ ) is obtained by inserting  $L - 1$  nulls between the individual samples of the input signal of the first sampling rate, where  $L$  is the factor by which the input signal ( $x(n)$ ) is up-sampled, and the first filtered up-sampled signal ( $y_L(n)$ ) is obtained by the modified convolution of the input signal ( $x(n)$ ) with the impulse response ( $h_{TP}$ ) of the Finite Impulse Response Filter according to

5

$$y_L(n) = \sum_{k=0}^{\lfloor \frac{N}{L} \rfloor} x(\lfloor n/L \rfloor - k) h_{TP, kL + \text{mod}(n, L)},$$

where L is the factor by which the input signal (x(n)) is up-sampled and N denotes the filter length of the low-pass filter.

- 10 **6.** The method according to claim 5, wherein the first output signal (v<sub>L</sub>(n)) of the second sampling rate is obtained by means of time-dependent filter coefficients (h<sub>TP</sub>) used for the low-pass filtering according to

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$$v_L(n) = a(n)y_L(n) + (1 - a(n))y_L(n - 1) = \mathbf{x}^T(n) \mathbf{h}_{TP}(n)$$

with

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$$\mathbf{x}(n) = \begin{bmatrix} x(\lfloor n/L \rfloor) \\ x(\lfloor n/L \rfloor - 1) \\ \dots \\ x(\lfloor n/L \rfloor - \lfloor N/L \rfloor) \end{bmatrix}$$

25

30

and

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$$\mathbf{h}_{TP}(n) = a(n) \begin{bmatrix} h_{TP, \text{mod}(n, L)} \\ h_{TP, L + \text{mod}(n, L)} \\ \dots \\ h_{TP, L \lfloor N/L \rfloor + \text{mod}(n, L)} \end{bmatrix} + (1 - a(n)) \begin{bmatrix} h_{TP, \text{mod}(n, L) - 1} \\ h_{TP, L + \text{mod}(n, L) - 1} \\ \dots \\ h_{TP, L \lfloor N/L \rfloor + \text{mod}(n, L) - 1} \end{bmatrix},$$

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where a(n) is the first weight factor and the upper index T denotes the transposed of a vector or a matrix.

- 45 **7.** The method according to one of the preceding claims, wherein the first output signal (v<sub>L</sub>(n)) of the second sampling rate is down-sampled (6) to obtain a second output signal (v(n)) of a third sampling rate different from the second sampling rate.

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- 8.** The method according to claim 7, wherein the first weight factor is given by

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$$a(n) = 1 - \left( \left[ \left[ n \frac{f_{in}}{f_{out}} - L \right] - n \frac{f_{in}}{f_{out}} - L + f_0 \delta_K(n) \right] \right)$$

with f<sub>0</sub> defined as

$$f_0 = \left( \left[ \frac{f_{in}}{f_{out}} L \right] - \frac{f_{in}}{f_{out}} L \right),$$

5 where  $\delta_K$  is the Kronecker-Delta function, L is the factor by which the input signal x(n) of the first sampling rate  $f_{in}$  is up-sampled,  $f_{out}$  is the third sampling rate of the second output signal (v(n)), and the symbols  $\lceil \cdot \rceil$  and  $\lfloor \cdot \rfloor$  denote rounding to the closest larger or smaller integer, respectively.

- 10 **9.** The method according to one of the claims 7 or 8, wherein the third sampling rate is time-dependent and the input signal (x(n)) is stored in a first buffer and/or the second output signal (v(n)) is stored in a second buffer.
- 15 **10.** The method according to claim 9, wherein the third sampling rate comprises a constant contribution and a time-dependent contribution given by

$$\Delta f_{out}(n) = \begin{cases} \min \left\{ \Delta f_{out}(n-1) + \Delta, \Delta f_{out,max} \right\}, & \text{if content of second buffer} \\ & \text{is below a first predetermined} \\ & \text{threshold} \\ \max \left\{ \Delta f_{out}(n-1) - \Delta, \Delta f_{out,min} \right\}, & \text{if content of second buffer} \\ & \text{is above a second predetermined} \\ & \text{threshold} \\ \Delta f_{out}(n-1), & \text{else} \end{cases}$$

35 and the second output signal (v(n)) is obtained by

$$\begin{aligned} 40 \quad v(n) &= \left( 1 - \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] + \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L - f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) \right) \\ &+ \left( \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] - \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L + f_0 \delta_K(n) \right) y_L \left( \sum_{m=0}^n R(m) - 1 \right) \end{aligned}$$

45 with

$$50 \quad R(n) = \left[ \sum_{i=0}^n \frac{f_{in}}{f_{out}(i)} L \right] - \sum_{m=0}^{n-1} R(m) + \left[ \frac{f_{in}}{f_{out}} L \right] \delta_K(n).$$

- 55 **11.** The method according to one of the claims 8 - 10, wherein the first sampling rate is time-dependent.

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12. Signal processing means, configured to generate from an input signal  $x(n)$  of a first sampling rate a first output signal  $v_L(n)$  of a second sampling rate that is different from the first sampling rate by carrying out processing steps according to one of the claims 1 - 6.

5 13. Signal processing means according to claim 12, configured to generate from an input signal  $x(n)$  of a first sampling rate a second output signal  $v(n)$  of a third sampling rate that is different from the first sampling rate and/or different from the second sampling rate by carrying out processing steps according to one of the claims 7-11.

10 14. Electrical device, in particular, a cellular phone or headset or hands-free set, comprising the signal processing means according to one of the claims 12 or 13.

15 15. Computer program product comprising one or more computer readable media having computer executable instructions for performing the steps of the methods of one of the claims 1-11.

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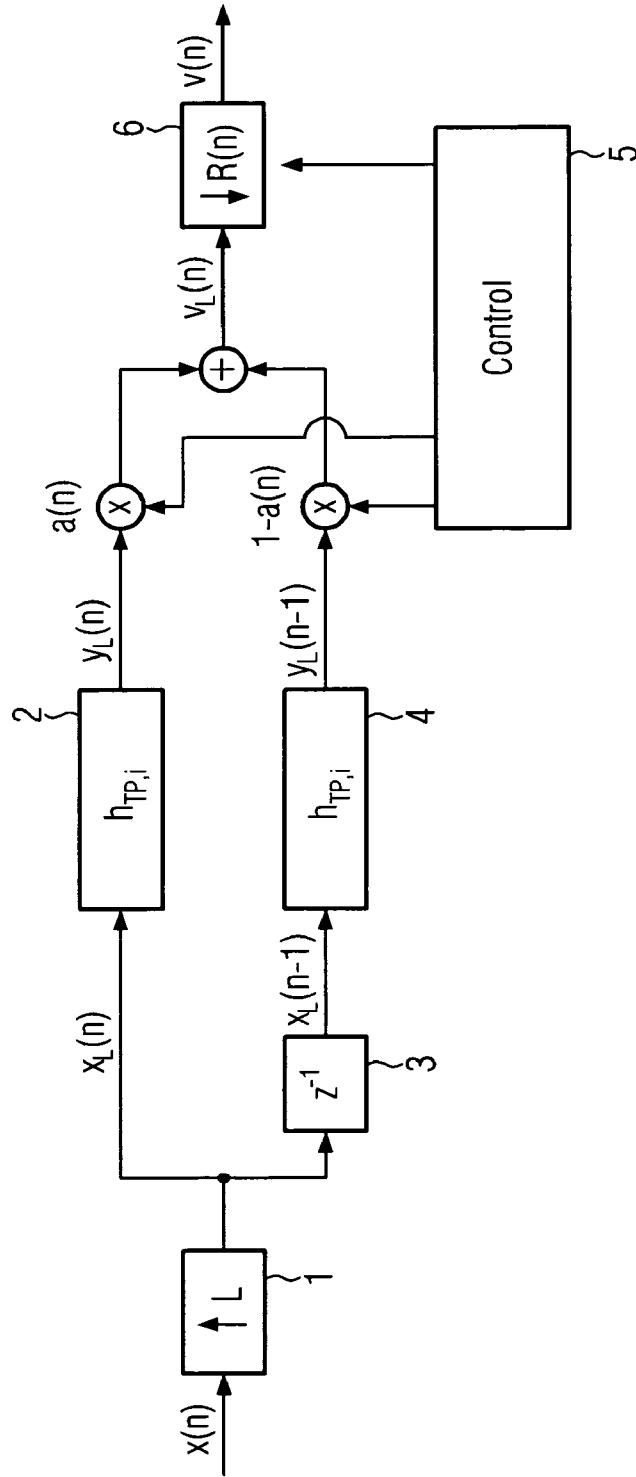


FIG. 1

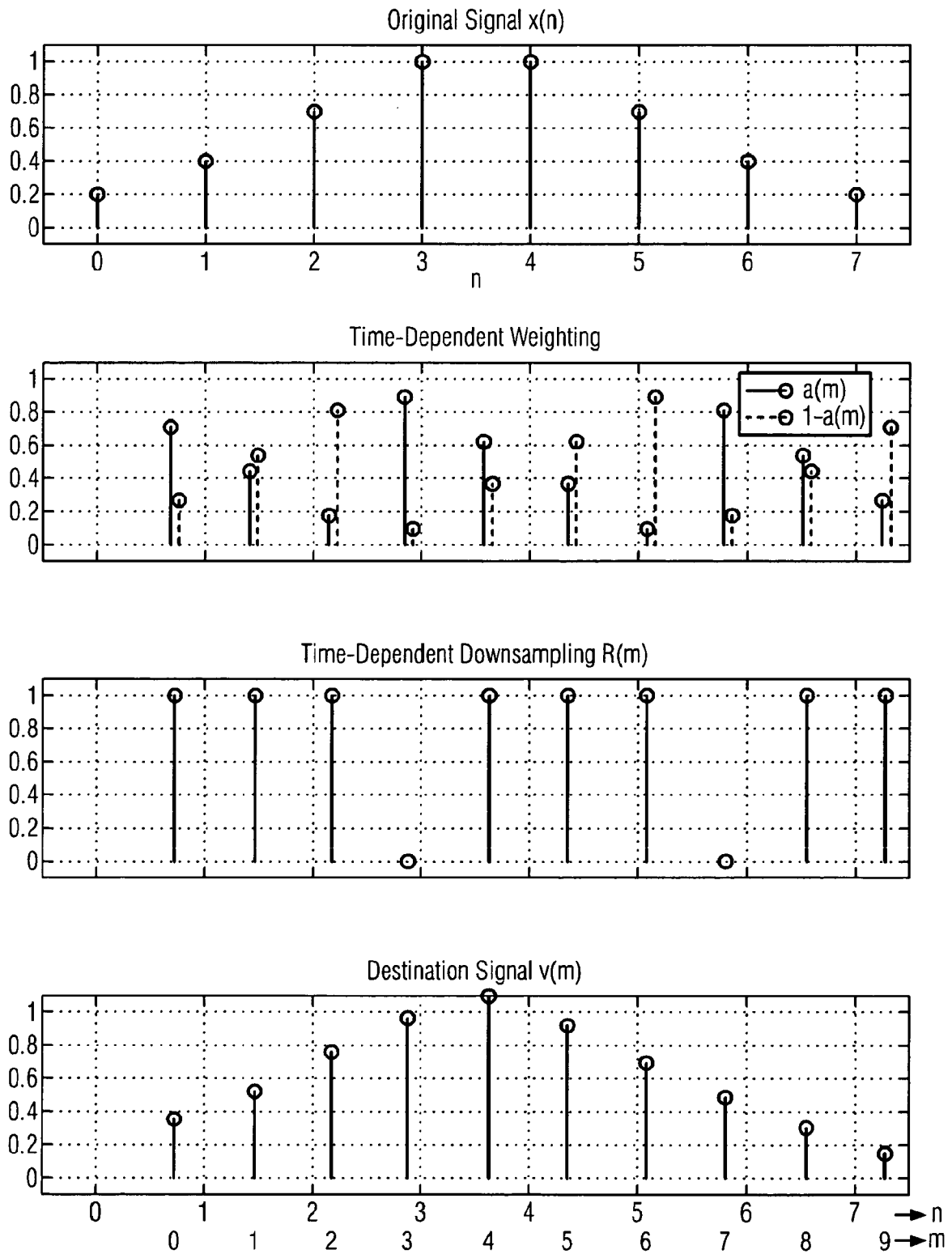


FIG. 2

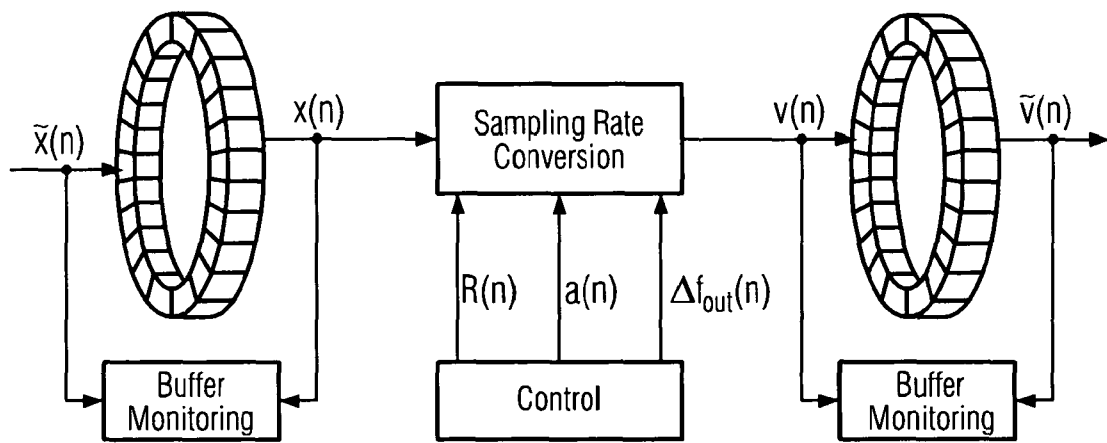


FIG. 3



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			H03H
The present search report has been drawn up for all claims			
Place of search The Hague		Date of completion of the search 27 June 2006	Examiner Lecoutre, R
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27-06-2006

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