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A flexible filter for synchronous angular resampling with a wireless sensor network

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ABSTRACT

Many tasks in the context of data acquisition with a wireless sensor network, e.g. angular resampling or jitter compensation, can be solved by reconstructing a uniformly sampled signal from a nonuniformly sampled one. A filter structure that is capable of performing this task has been adopted from the literature. It has been implemented in fixed-point arithmetic on a wireless sensor node. Compared to a floating-point implementation, this fixed point implementation has been shown to provide the same signal quality at only about one-tenth of computation time. Furthermore, it has been proven to be suitable for real-time operation on a resource constraint wireless sensor node. The implementation has been tested for the application example of synchronous angular resampling, but it can also easily be used in other resampling applications. Experiments with simulated data, a waveform generator and at an induction motor test bench show that the implemented filter is effective in resampling signals over a wide range of speeds and speed changes. In all cases its signal quality in terms of amplitude and frequency accuracy is at least as good that obtained using sample-and-hold resampling and surpasses it clearly in terms of signal-to-noise and distortion ratio (SINAD).

1. Introduction

The resampling of data that has been obtained on a nonuniform sampling grid to a uniform one can be used to solve various problems in data acquisition with wireless sensor networks. Firstly, it has been argued in [1] that in order to obtain a set of synchronized samples from a wireless sensor network, it may be beneficial to sample signals in an unsynchronized way first and resample them to a synchronized sampling grid later. This approach has the advantages that it matches better with irregular network communication patterns and helps to reduce interdependencies between different software modules [1]. Due to clock jitter or non-linear drift of the unsynchronized sampling clock, the unsynchronized samples, in general, will have been obtained on a nonuniform sampling grid [1]. Thus, resampling to the usually uniform global sampling grid involves the recovery of a uniformly sampled signal from a nonuniform set of samples. Another application where a nonuniform set of samples has to be transformed to a uniform one, is the resampling of a signal that has been sampled uniformly in time to a signal that is sampled uniformly with reference to the rotation angle of a machine. This is a common procedure in machine diagnosis [2]. The frequencies of the signals sampled over the angle is commonly described in orders, where the frequency of an n-th order signal is n-times that of the rotation rate. Very similar to that is the removal of doppler shifts from acquired signals [2].

This article is an extended version of our paper [3] presented at the 20th IMEKO TC-4 International Symposium. It gives a more detailed introduction into the theory of nonuniform sampling and the filter structure that is used. Furthermore, the description of a practical implementation on a commercial wireless sensor node and experimental results obtained with it have been added.

The organization of this article is as follows: Section 2 gives an overview of existing algorithms for synchronous angular resampling and general nonuniform resampling algorithms. An introduction into the theory of nonuniform sampling and the used filter structure is given in Section 3. The design and implementation of the filter structure on a commercial wireless sensor node is described in Section 4. Experiments to characterize the implemented filter are presented in Section 5. The corresponding results are discussed in Section 6. Finally, the conclusions and an outlook on future research are given in Section 7.

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2. Related work

Synchronous resampling is a well established technique in machine diagnosis [2], but new in the synchronous data acquisition with wireless sensor networks [1].

In [4] various methods for nonuniform reconstruction are reviewed. Amongst those, low-pass filtering and polynomial interpolation seem to be the most widely used. By low-pass filtering a modulated delta comb, a continuous signal can be reconstructed out of nonuniform samples. This is the same method that is usually used to reconstruct continuous signals form uniform samples. Resampling the so generated continuous signal at uniform time instants completes the reconstruction process (used e.g. in [5]). However, this method does not remove multiplicative noise from the signal that is generated during nonunifrom sampling [6] (compare (6) in Section 3.1).

Another method frequently used is polynomial interpolation [4]. It is computationally very simple, especially for low polynomial orders. Yet, at low polynomial orders the usable signal bandwidth and alias attenuation are both very low [7]. At higher polynomial orders the computational complexity quickly increases [4]. Thus, polynomial interpolation is restricted to be used with highly oversampled narrow-band signals.

In [8] nonuniform sample reconstruction is used to reconstruct lost samples of speech signals. The method that is used is derived from an analysis of the spectra of a nonuniformly sampled signal and the corresponding delta comb. Thus, it is a theoretically well founded method for general nonuniform reconstruction [9]. Later in [6], an efficient digital filter based on this method for the use in radio communication was presented.

In [2] two methods for angular resampling are presented. The first uses a FIR interpolation filter to upsample the input signal by a constant integer factor. The upsampled signal is then resampled to the correct angular instants using linear interpolation. Finally, the resampled signal is downsampled using an FIR decimator. The FIR interpolation filter ensures that the signal is sufficiently band-limited to be safely used with the linear interpolator. This method is effective for angular resampling if the supported rpm-range is small. At higher rpm-ranges it requires increasingly narrow-band and high order FIR filters leading to a tremendous computational complexity.

The second method presented in [2] applies the hybrid model for resampling (compare Fig. 2) to a continuous filter whose impulse response is a truncated sinc-function. The values of this impulse response are stored in a look-up table which can be of considerable size (≈ 8192 Elements). Again this method is quite efficient for low rpm-ranges. But at high rpm-ranges the required narrow filter bandwidth resulting in a wide impulse response makes it computationally infeasible, as well.

In conclusion, a number of methods for nonuniform reconstruction as well as synchronous angular resampling can be found in the literature. However, none of those has been implemented on a wireless sensor network so far. Most methods, either do not provide satisfactory results or are computationally to expensive to be run on a resource constraint wireless sensor node. Currently, the method from [8,9] in the form described in [6] seems to be the most promising for the use in a wireless sensor network, as it is theoretically well founded, offers a good signal quality in a wide range of applications and can be implemented in a computationally efficient way.

3. A filter for uniform sample recovery

This section introduces a filter that can be used to recover uniform samples from a set of nonuniform ones. Section 3.1 outlines the general spectrum of a nonuniformly sampled signal and a reconstruction method that can be deduced from that. A realization of this method as a digital filter is described in Section 3.3. Section 3.2 introduces the transposed Farrow filter. It can be used to resample signals at arbitrary time instants and is a key component of the reconstruction filter.

3.1. Spectrum of a nonuniformly sampled signal

According to [8,9] the signal x_s sampled at the nonuniform time instants t_k can be modeled as the product of the original signal x and the comb function x_p:

$$x_s(t) = x(t)x_p(t) \tag{1}$$

$$x_p(t) = \sum_{k=-\infty}^{\infty} \delta(t - t_k) \tag{2}$$

where δ is the dirac delta function. The fourier series expansion of x_p yields [9]:

$$x_p(t) = \frac{\left|1 - \dot{\Theta}(t)\right|}{T}\left[1 + 2\sum_{k=1}^{\infty}\cos\left(\frac{2\pi kt}{T} - \frac{2\pi k\Theta(t)}{T}\right)\right] \tag{3}$$

$$x_s(t) = x(t)\frac{\left|1 - \dot{\Theta}(t)\right|}{T}\left[1 + 2\sum_{k=1}^{\infty}\cos\left(\frac{2\pi kt}{T} - \frac{2\pi k\Theta(t)}{T}\right)\right] \tag{4}$$

where T is the uniform sampling interval and $\Theta(t_k) = t_k - kT$ the deviation from the uniform sampling instants. It is noted in [9] that x_p is essentially a phase modulated signal. Fig. 1 shows an example from [9] of the spectrum of a series of nonuniform sampling pulses x_p.

If the bandwidth W_Θ of $\Theta(t)$ is less than $\frac{1}{2T}$, the modulation spectra do not overlap with the base band component [9]. Thus, is can be recovered by lowpass filtering [9]:

$$x_{p,lp}(t) = \frac{\left|1 - \dot{\Theta}(t)\right|}{T} \tag{5}$$

$$x_{s,lp}(t) = x(t)\frac{\left|1 - \dot{\Theta}(t)\right|}{T} \tag{6}$$

Now the original signal can be recovered as [9]:

$$\tilde{x}(t) = \frac{x_{s,lp}(t)}{x_{p,lp}(t)} \approx x(t) \tag{7}$$

3.2. Transposed farrow filter

The transposed farrow filter [5] implements the hybrid filter model shown in Fig. 2. Conceptually, the discrete signal $x[k]$ is used to modulate a delta comb resulting in a continuous signal $x(t_k)$. This signal is then filtered using a continuous filter with the impulse response h_a. Finally, the filter output is sampled at the time instants lT_{out} producing the resampled discrete signal $y[l] = y(lT_{out})$.

This hybrid model can be implemented by a completely discrete filter structure, as described in [5]: The convolution integral (8) of the continuous filter can be reduced to a sum (9), as it has only discrete in- and outputs:

$$y(t) = \int_{-\infty}^{\infty} x(\tau)h_a(t - \tau)\,\mathrm{d}t \tag{8}$$

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